IP Network Engineering Guidelines

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Reference list

The following are the references for this section:

- *Software Input/Output: Administration* (553-3001-311)
- *Data Networking Guidelines* (553-3023-103)

Overview

This chapter provides guidelines and recommendations to help plan, engineer, and test the Voice Gateway Media Card and Internet Telephone network on the Meridian 1 system.

For Succession Communication Server for Enterprise (CSE) 1000 Release 2.0 systems, also refer to *Data Networking Guidelines* (553-3023-103).

The following procedures are contained within this chapter:

- Procedure 1, "Performing the Network assessment procedure" on page 138
- Procedure 2, "Calculating TLAN traffic" on page 156
- Procedure 3, "Calculating WAN traffic" on page 161
- Procedure 4, "Assessing link utilization" on page 167

See "Configuration of the DHCP Server" on page 211, for engineering guidelines to set up and configure the Dynamic Host Configuration Protocol (DHCP) server to support the Voice Gateway Media Card and Internet Telephones.

IP address requirements

This section describes the IP address requirements for each node, for each card, and for each Internet Telephone.

A node is a group of ITG-P Line Cards and Succession Media Cards within a given Meridian 1 or Succession CSE 1000 system. Each card within a node has two IP addresses: a node for the Telephony LAN (TLAN) and a node for the Meridian 1 or Succession CSE 1000 Embedded LAN (ELAN). Each node has one Node IP address on the TLAN, that is dynamically assigned to the connection server on the node Master. The Internet Telephone uses the Node IP address during the registration process.

All ELAN addresses for all nodes must be on one subnet. All ELAN addresses must be on the same subnet as the Meridian 1 or Succession CSE 1000 Core ELAN. All TLAN addresses must be in the same subnet for a given node.

General Requirements for a Node's IP Addressing

The following is a list of IP addresses that must be assigned to configure a node:

- IP address for every TLAN interface of every Voice Gateway Media Card
- IP address for every ELAN interface of every Voice Gateway Media Card
- Voice LAN (TLAN) Node IP address (This address is shared among all the cards.) This alias IP address appears dynamically on the TLAN port of one card in the node, the Leader or node Master.
- On the Succession CSE 1000 Rel 2.0 system, an IP address for the Signaling Server ELAN and Signaling Server TLAN

In addition to the IP addresses that must be assigned, additional network information must be entered:

- Management LAN (ELAN) gateway IP address
- Management LAN (ELAN) subnet mask
- Voice LAN (TLAN) subnet mask
- VLAN gateway IP address

CAUTION
You must use separate subnets with the Voice Gateway Media Card for ELAN and TLAN.

The default setting of separate ELAN and TLAN subnets offers the following benefits:

- Separate subnets are easier to configure for traffic management and Quality of Service (QoS).
- Separate subnets protect the Meridian 1 and Succession CSE 1000 ELAN from general LAN traffic, including broadcast and multicast storms
- Separate subnets are more secure against unauthorized access

Voice Gateway Media Card IP address requirements

The IP address information for each card is set in the **Configuration** tab of the **ITG Node Properties** window of the IP Telephony Gateway - IP Phones application. The IP address requirements for each card depend on the node subnet option.

You must provide an IP address for an ELAN and TLAN port. On the ELAN, all cards must be on the same subnet, which is the same subnet that the Meridian 1 and Succession CSE 1000 is connected to. On the TLAN, all cards in a node must be on the same subnet.

The ELAN address corresponds to the Management MAC address which is assigned during manufacturing and cannot be changed. Locate the faceplate sticker on the Voice Gateway Media Card. The ELAN/Management MAC address is the MOTHERBOARD Ethernet address.

Separate subnet Voice Gateway Media Card IP address requirements

You must use separate subnets for the IP Telephony node. Each Voice Gateway Media Card requires a:

- Management IP address
- Voice IP address
- Management MAC
- Voice LAN gateway IP address

IP network assessment procedure

An efficient IP Line network design begins with an understanding of traffic and the underlying network that carries the traffic. To determine the network requirements of the specific system, the technician must perform the steps in Procedure 1.

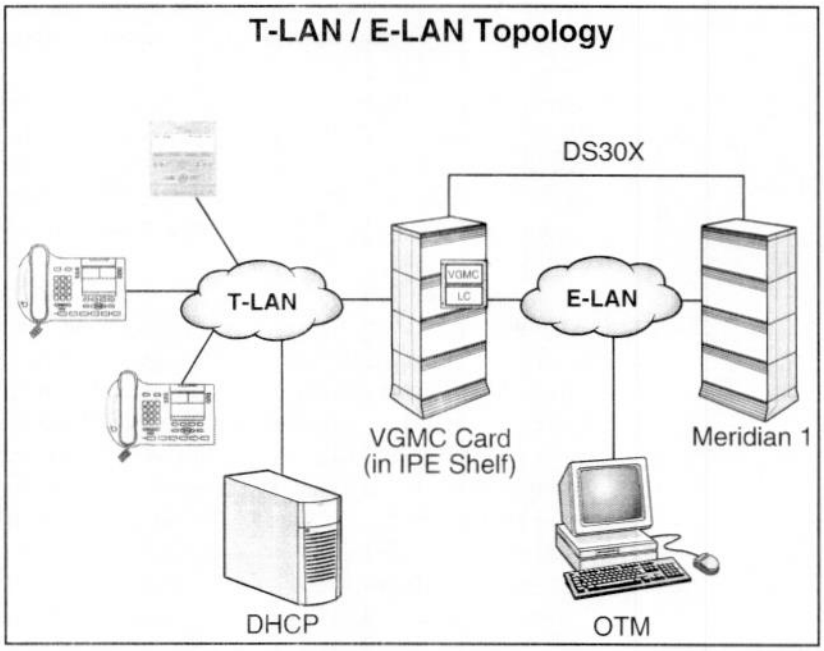
Procedure 1
Performing the Network assessment procedure

1. Estimate the amount of traffic processed by the Meridian 1 or Succession CSE 1000 system through the IP Line network. See "Calculate IP Line traffic requirements" on page 155.
2. Assess if the existing corporate intranet can adequately support voice services. See "Calculate IP Line traffic requirements" on page 155 and "Assess WAN link resources" on page 161.
3. Organize the IP Line network into "zones" representing different topographical areas of the network that are separated according to bandwidth considerations. See "VoIP bandwidth management zones" on page 164.
4. Set a variety of service parameters to improve service and coordinate (with the IP administrator) the prioritization of voice packets with data traffic. See "Set service parameters" on page 171.
5. Provide the necessary IP network infrastructure:
   a. 10BaseT or 100BaseTX Ethernet connection.
   b. IP address. Each Voice Gateway Media Card requires 10BaseT ELAN or 10/100BaseT TLAN unicast IP address.
   c. One additional IP address for each node. The node IP address is the TLAN for a subnet.

End of Procedure

After completing the network assessments, the technician can design and implement the IP Line network. This can involve modifications to both the IP Line elements and to the existing network. Post-installation network measurements (see page 191) must be made on a regular basis to make sure QoS standards are maintained. Figure 6 shows an example of the TLAN and ELAN topology.

Figure 6
TLAN and ELAN Topology



Codecs

The Internet Telephones and Voice Gateway Media Cards support different codecs and codec parameters with different compression rates and audio quality. The Meridian 1 and Succession CSE 1000 selects the appropriate codecs based on user-configurable parameters. For instance, an Internet Telephone-to-Internet Telephone within a LAN can be set up using G.711 at 64 Kbps. For an Internet Telephone-to-Internet Telephone call over a WAN, the call can be set up using G.729A or G.729AB at 8 Kbps. These data rates and the Voice Gateway Channel Server on the Voice Gateway Media Card are for the voice stream only. Packet overhead is not included.

The Terminal Proxy Server (TPS) and the Voice Gateway Channel Server on the Voice Gateway Media Card have a predefined table of codec option sets that can be supported. The first entry in the table has the highest quality audio (BQ = Best Quality) and requires the largest bandwidth. The last entry requires the least bandwidth (BB = Best Bandwidth) with lower voice quality.

When the Call Server sets up a Call Server connection between an Internet Telephone-to-Internet Telephone or Internet Telephone-to-Voice Gateway Channel Server, the predefined table determines which codec it selects for that connection. This information is provided to the Meridian 1 and Succession CSE 1000 as part of the Internet Telephone registration sequence. For more information about the registration sequence, refer to "Configuration of the DHCP Server" on page 211. The Meridian 1 and Succession CSE 1000 use this information to set up a speech path to select a codec that both endpoints support. As part of zone management, it further selects the codec based on whether it is trying to optimize quality (BQ) or bandwidth (BB).

CAUTION
When voice compression codecs are used, voice quality is impaired if end-to-end calls include multiple compressions.

Codec refers to the voice coding and compression algorithm used by the DSPs on the Voice Gateway Media Card. Different codecs provide different levels of voice quality and compression properties. The specific codecs and the order in which they are used are configured in the TPS, and Meridian 1 and Succession CSE 1000.

Table 32 shows which codecs are supported on the Meridian 1 and Succession CSE Rel 1.1 systems, and on the Succession CSE 1000 Rel 2.0 systems:

Table 32
Supported codecs

Codec	Supported on Meridian 1 and Succession CSE 1000 Rel 1.1	Supported on Succession CSE 1000 Rel 2.0
G.711	Yes	Yes
G.729A	Yes	Yes
G.729AB	Yes	Yes
G.723.1	No	Yes
T.38	No	Yes
G.711 Clear Channel	No	Yes

Note 1: The G.723.1 codec is supported only on Succession CSE 1000 Release 2.0 system. The supported G.723.1 codec has bit rates of 5.3 Kbps and 6.3 Kbps. The user can configure the G.723.1 codec with a 5.3 Kbps bit rate; however, the system accepts both G.723.1 5.3Kbps and 6.4Kbps from the far end.

Note 2: The T.38 and G.711 Clear Channel codecs are supported for fax calls and are only supported on the Succession CSE 1000 Release 2.0 system. T.38 is the preferred codec type for fax calls over virtual trunks. However, the G.711 Clear Channel codec is used if the far end does not support the T.38 codec.

Table 33 lists the payload sizes for the different codecs:

Table 33
Codec payload sizes

Codec	Payload
G.711	10ms, 20ms, 30ms
G.729A	10ms, 20ms, 30ms, 40ms, 50ms
G.729AB	10ms, 20ms, 30ms, 40ms, 50ms
G.723.1	30ms
T.38	supported for fax calls
G.711 Clear Channel	supported for fax calls

Note: If there are multiple nodes on a system and the same codec is selected on more than one node, ensure that each node has the same voice payload size configured for the codec.

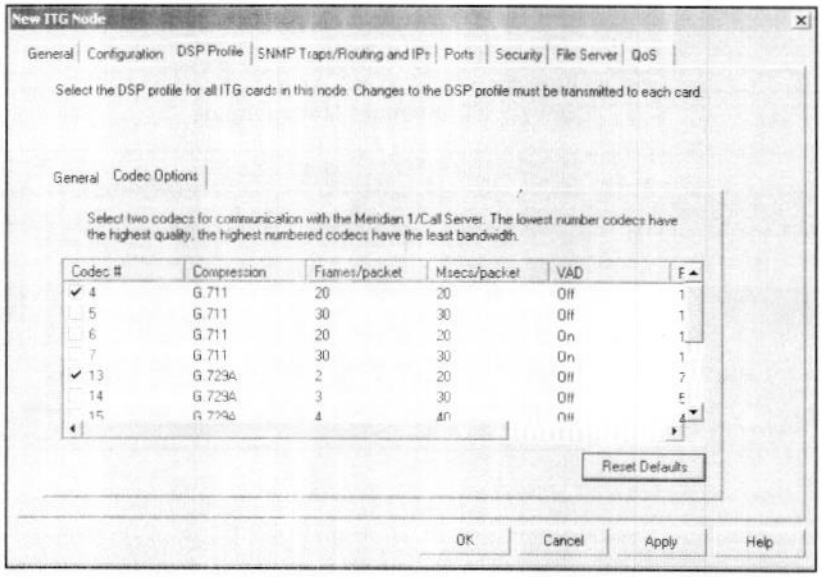
Codec configuration

Configure codec in the DSP Profile sections of OTM and Element Management.

Meridian 1 and Succession CSE 1000 Rel 1.1

Figure 7 on page 143 shows the list of codecs available on the DSP Profile tab within OTM's ITG Line 3.0 application. The Codec Options sub-tab presents a table of different sets of codec options identified by a codec setting index number. There is a list of up to 32 codec settings for G.711, G.729A, and G.729AB. The lesser codec setting index corresponds to BQ (Best Quality) in LD 117 zone configuration. The greater codec setting index corresponds to BB (Best Bandwidth). For more information, see"Codec selection" on page 191.

Figure 7
Codec list on OTM 2.0

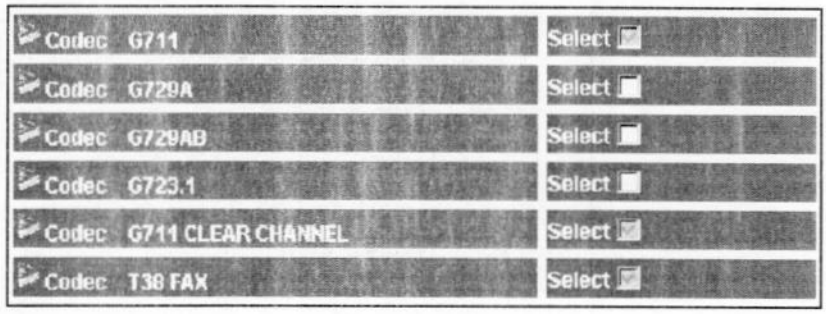


For codec configuration in the Meridian 1 and Succession CSE 1000 Rel 1.1 systems, see "Configuring DSP profile data" on page 284.

Succession CSE 1000 Rel 2.0

Figure 8 on page 144 shows the list of codec types that are displayed in the Element Management.

Figure 8
Codec list in Element Management



The G.711, G.711 Clear Channel, and T.38 Fax codecs are automatically selected and cannot be unselected. Even though these codecs cannot be unselected, the payload size, and the jitter buffer for G.711 can be changed. For G.711 Clear Channel, only the jitter buffer can be changed.

The user can select all three, any two, any one, or none of the G.729A, G.729AB, and G.723.1 codecs. If the G.729A or G.729AB codec is selected, the user can change the payload size and the jitter buffer. If the G.723.1 codec is selected, the user can change only the jitter buffer, because the only supported payload size is 30 msec.

For codec configuration in the Succession CSE 1000 Rel 2.0 system, see "Configuring DSP Profile data" on page 346.

Codec Registration on Succession CSE 1000 Rel 2.0

After the configuration of codecs is complete, the Internet Telephones and DSPs have to register the configured codecs with the Call Server.

Codec registration for Internet Telephones

The Internet Telephones register both the G.711 a-law and mu-law codecs, and any codec(s) configured by the user. The codecs that can be configured by the user are G.729A, G.729AB, and G.723.1.

> *Note:* Internet Telephones do not register the fax codecs (T.38 and G.711 Clear Channel).

The minimum number of codecs registered for an Internet Telephone is two: G.711 a-law and G.711 mu-law (G.711 is always configured).

The maximum number of codecs registered for an Internet Telephone is five: G.711 a-law, G.711 mu-law, G729A, G729AB, and G.723.1.

Example 1

A user configures a G.711 mu-law codec (with a 30 msec payload) and a G.723.1 codec (with a 30 msec payload).

The following three codecs are actually registered:

1 G.711 mu-law (30 msec)

2 G.711 a-law (30 msec)

3 G.723.1 (30 msec)

Example 2

A user configures four codecs:

1 G.711 a-law codec with a 10 msec payload

2 G.729A codec with 50 msec payload

3 G.729AB codec with 30 msec payload

4 G.723.1 codec with a 30 msec payload

The following five codecs are actually registered:

1 G.711 a-law (10 msec)

2 G.711 mu-law (10 msec)

3 G.729A (50 msec)

4 G.729AB (30 msec)

5 G.723.1 (30 msec)

Codec registration for DSPs

DSPs register the following codecs:

- both G.711 a-law and G.711 mu-law codecs
- both fax codecs (T.38 and G.711 Clear Channel)
- one best bandwidth (BB) codec if at least one of G.729A, G.729AB, or G.723.1 codecs was configured by the user. The best bandwidth (BB) codec is based on the codec type. The order of preference for choosing the best bandwidth codec is G.729AB, G.729A, and then G.723.1.

The minimum number of codecs registered for DSPs is four: G.711 a-law, G.711 mu-law, T.38 and G.711 Clear Channel.

The maximum number of codecs registered for DSPs is five: G.711 a-law, G.711 mu-law, T.38 and G.711 Clear Channel and one of G.729AB, G.729A, or G.723.1.

Example 1

A user configures four codecs:

1 G.711 a-law codec with a 10 msec payload

2 G.729A codec with 50 msec payload

3 G.729AB codec with 30 msec payload

4 G.723.1 codec with a 30 msec payload

The following five codecs are actually registered:

1 G.711 a-law (10 msec)

2 G.711 mu-law (10 msec)

3 G.729AB (30 msec)

4 T.38

5 G.711 Clear Channel

The G.729AB codec is selected, as it is the first in the order of preference of the "best bandwidth" codecs. The G.729A and G.723.1 codecs do not get registered.

Example 2

A user configures three codecs:

1 G.711 mu-law codec with a 20 msec payload

2 G.729A codec with 30 msec payload

3 G.723.1 codec with a 30 msec payload

The following five codecs are actually registered:

1 G.711 mu-law (20 msec)

2 G.711 a-law (20 msec)

3 G.729A (30 msec)

4 T.38

5 G.711 Clear Channel

The G.729A codec is selected, as it precedes the G.723.1 codec in the order of preference of the "best bandwidth" codecs.

Codec negotiation for Succession CSE 1000 Rel 2.0

For every virtual trunk call, a common codec must be selected for the call. This is known as codec negotiation. Codec negotiation for virtual trunk calls is performed through the H.323 FastStart and Terminal Capability Set (TCS) messages.

For a call setup with the FastStart procedure, the originating node sends its codec list in the FastStart element in the SETUP message to the terminating node. For a call setup using the SlowStart procedure or for a call modification (media redirection), each node sends its codec list in the TCS message to the other node.

Before sending a codec list in FastStart and TCS messages, the codec list must be sorted according to the Best Bandwidth or Best Quality policy. This is determined by the following:

- the zone configuration of the Internet Telephone / DSP involved in the call
- the zone configuration of the virtual trunk used for the call

Codec list sorting

There are two methods for sorting the codec list:

1 Best Quality (BQ) sorting - The codec list is sorted so that the first codec in the list is the best quality codec, the second codec is the second best quality codec in the list, and so on.

2 Best Bandwidth (BB) sorting - The codec list is sorted so that the first codec in the list is the best bandwidth codec, the second codec is the second best bandwidth codec in the list, and so on.

Table 34 on page 149 shows the codec list sorting order for the BQ and BB codecs. To know if a codec is best quality (as compared to another codec), refer to the lists in columns 1 and 2. To know if a codec is best bandwidth (as compared to another codec), refer to the lists in columns 3 and 4. The best quality or bandwidth codec is listed at the top of the column.

Table 34
BQ and BB codec sorting lists

Best Quality (BQ) Sorting		Best Bandwidth (BB) Sorting	
For mu-law systems	**For a-law systems**	**For mu-law systems**	**For a-law systems**
G.71_mu_law_10msec	G.711_a_law_10msec	G.729AB_50msec	G.729AB_50msec
G.711_mu_law_20msec	G.711_a_law_20msec	G.729AB_40msec	G.729AB_40msec
G.711_mu_law_30msec	G.711_a_law_30msec	G.729AB_30msec	G.729AB_30msec
G.711_a_law_10msec	G.711_mu_law_10msec	G.729AB_20msec	G.729AB_20msec
G.711_a_law_20msec	G.711_mu_law_20msec	G.729AB_10msec	G.729AB_10msec
G.711_a_law_30msec	G.711_mu_law_30msec	G.729A_50msec	G.729A_50msec
G.729A_10msec	G.729A_10msec	G.729A_40msec	G.729A_40msec
G.729A_20msec	G.729A_20msec	G.729A_30msec	G.729A_30msec
G.729A_30msec	G.729A_30msec	G.729A_20msec	G.729A_20msec
G.729A_40msec	G.729A_40msec	G.729A_10msec	G.729A_10msec
G.729A_50msec	G.729A_50msec	G.723.1_5.3kbps_30ms	G.723.1_5.3kbps_30ms
G.729AB_10msec	G.729AB_10msec	G.723.1_6.4kbps_30ms	G.723.1_6.4kbps_30ms
G.729AB_20msec	G.729AB_20msec	G.711_mu_law_30msec	G.711_a_law_30msec
G.729AB_30msec	G.729AB_30msec	G.711_mu_law_20msec	G.711_a_law_20msec
G.729AB_40msec	G.729AB_40msec	G.711_mu_law_10msec	G.711_a_law_10msec
G.729AB_50msec	G.729AB_50msec	G.711_a_law_30msec	G.711_mu_law_30msec
G.723.1_5.3kbps_30ms	G.723.1_5.3kbps_30ms	G.711_a_law_20msec	G.711_mu_law_20msec
G.723.1_6.4kbps_30ms	G.723.1_6.4kbps_30ms	G.711_a_law_10msec	G.711_mu_law_10msec
T.38	T.38	T.38	T.38
G.711CC	G.711CC	G.711CC	G.711CC

Codec selection for Succession CSE 1000 Rel 2.0

For every virtual trunk call, a codec must be selected before the media path is opened.

When a call setup with the FastStart procedure is used, the terminating node selects a common codec and sends the selected codec to the originating node. For a call modification (media redirection) or for a call setup using the SlowStart procedure, the codec selection occurs on both nodes. Each node has two codec lists: its own list and the far-end's list. To select the same codec on both nodes, it is essential to use the same codec selection algorithm on both nodes.

For the codec selection, both the near- and far-end codec lists are retrieved:

- The far-end list is not modified because it is already sorted when it is received (in FastStart or TCS message).
- The near-end list is sorted and then expanded to include lower payloads, the same way it is done before sending the codec list in FastStart message.

The following conditions are met before codec selection occurs:

- There are two codec lists:
  - The near-end list is the codec list of the local unit.
  - The far-end list is the codec list received from the far end.
- Each codec list can contain more than one payload size for a given codec type. The codec list depends on the codec configuration.
- Each codec list is sorted by order of preference. The first codec in the near end list is the near end's most preferred codec and the first codec in far end list is the far end's most preferred codec, and so on.

Once the above conditions are met, a codec selection algorithm is used to select the codec to be used for a call. There are two different codec selection algorithms:

1 H.323's Master/Slave algorithm

2 Best Bandwidth Codec Selection algorithm

H.323's Master/Slave algorithm

The codec selection algorithm proposed by the H.323 standard involves a Master/Slave negotiation, initiated each time two nodes exchange their capabilities (TCS message). The Master/Slave information decides that one node is Master and the other node is Slave. The outcome of the Master/Slave negotiation is not known in advance, it is a random result: one node could be Master then Slave (or vice versa) during the same call.

- The Master node uses its own codec list as the preferred one. From the far-end list, it finds the common codec.

  The Master gets the first codec in its own list (Codec1). The Master then checks the far-end list to see if Codec1 is a common codec (that is, is Codec1 also listed in the far-end list). If Codec1 is common to both lists, Codec1 becomes the selected codec. Otherwise, the Master obtains the second codec from its own list and repeats the search in the far-end list, and so on.

- The node which is Slave uses the far end list as the preferred list. The Slave selects a codec from the far end list and then searches in its own list to find the common codec.

The issues caused by the Master/Slave algorithm are due to the random nature of the Master/Slave information. The codec that is selected and used during a virtual trunk call cannot be pre-determined. This issue can make bandwidth usage calculations and bandwidth management difficult.

Known issues include:

- After an on-hold and off-hold scenario (that triggers Master/Slave negotiation), the codec used for the restored call can be different than the codec used before the call was placed on hold. The Master/Slave information could have been changed when the call was on hold.
- Since the terminating end of a call is always the Master, a call from telephone1 (node1) to telephone2 (node2) can use a different codec than a call from telephone2 (node2) to telephone1 (node1).
- For tandem calls, the Master/Slave information is not relevant. That is, the Master/Slave information is designed to be used only between two nodes, not among three or more nodes. The Master/Slave algorithm makes the codec selection for tandem calls more complex and inefficient.

To solve the issues, another codec selection algorithm was needed. This new algorithm is called the Best Bandwidth Codec Selection algorithm and it is not based on the unpredictable Master/Slave information.

The new codec algorithm is used for virtual trunk calls between Nortel Networks equipment, since any change to the Master/Slave algorithm implies a change to the H.323 standard. However, the H.323's Master/Slave algorithm is used when there is a virtual trunk call between Nortel Networks equipment and third-party equipment.

Best Bandwidth Codec Selection algorithm

The Best Bandwidth Codec Selection algorithm was implemented to solve the issues caused by the H.323 Master/Slave algorithm. The Best Bandwidth algorithm selects one common codec based on two codec lists. With this algorithm, every time the selection is done using the same two lists, the selected codec is always the same.

The "Best Bandwidth" codec selection is based on the codec type only; it does not take into account the fact that some codecs, while generally using less bandwidth, consume more bandwidth than others at certain payload sizes.

- This algorithm obtains the first codec in the near-end list that is also in far end list (codec is the same type and has the same payload size). Call the selected codec C1.
- Get the first codec in far end list that is also in near-end list (same type, same payload size). This codec is C2.
- Between C1 and C2, the codec that is selected is considered as the best bandwidth codec type. To determine which codec type is the best bandwidth, the following rules are used:
  - a G.729AB codec is considered as Best Bandwidth compared to G.729A, G.723.1, G.711_muLaw, and G.711_aLaw codecs
  - a G.729A codec is considered as Best Bandwidth compared to G.723.1,G.711_muLaw, and G.711_aLaw codecs
  - a G.723.1 codec is considered as Best Bandwidth compared to a G.711_muLaw and G.711_aLaw codec
  - a G.711_muLaw codec is considered as Best Bandwidth compared to a G.711_aLaw codec

Table 35 shows the codec that would be selected between any two codecs. For example, if the two codecs are the G.729A and G.723.1, the selected codec is the G.729A.

Table 35
Best Bandwidth codec selection between any two codecs types

Codec type	G.711_aLaw	G.711_muLaw	G.729A	G.729AB	G.723.1
G.711_aLaw	G.711_aLaw	G.711_muLaw	G.729A	G.729AB	G.723.1
G.711_muLaw	G.711_muLaw	G.711_muLaw	G.729A	G.729AB	G.723.1
G.729A	G.729A	G.729A	G.729A	G.729AB	G.729A
G.729AB	G.729AB	G.729AB	G.729AB	G.729AB	G.729AB
G.723.1	G.723.1	G.723.1	G.729A	G.729AB	G.723.1

Calculate IP Line traffic requirements

The technician must forecast the hundreds of call seconds for each hour (CCS) traffic that the Meridian 1 and Succession CSE 1000 processes through the IP Line network. CCS traffic generated by an Internet Telephone is similar to that of a digital telephone. The following procedures calculate the bandwidth required to support given amounts of traffic.

The procedures require the following data:

- CCS/CCS rating of Internet Telephone
- number of Internet Telephones
- number of subnets/servers accessed by the Internet Telephones

Note: Base all traffic data on busy hour requirements.

The result of the calculation provides estimated values for the following:

- total TLAN bandwidth requirement
- WAN bandwidth requirement for each subnet or server/router

The technician must consider the impact of incremental IP Line traffic on routers and LAN resources in the intranet. LAN segments can become saturated, and routers can experience high CPU use. A customer must consider re-routing scenarios in a case where a link is down.

TLAN traffic calculations

To calculate the total TLAN requirement, add together all sources of traffic destined for the Internet Telephony network using the same LAN. The data rate for a TLAN is the total bit rate. The total subnet traffic is measured in Erlangs. An Erlang is a telecommunications traffic measurement unit and it is used to describe the total traffic volume of one hour. Network designers use these measurements to track network traffic patterns. To calculate the TLAN traffic, follow Procedure 2.

Procedure 2
Calculating TLAN traffic

1 Total subnet traffic is the sum of (measured in Erlangs):

- number of Internet Telephones × CCS/CCS rating)
- voice gateways on Voice Gateway Media Card
- WAN connection

Note: Each source of traffic has a different CCS rating. Calculate the subnet traffic for each source of traffic and add the amounts to get the total.

2 Use the number of Erlangs to calculate the equivalent number of lines by using the calculator at the following Web site:

http://www.erlang.com/calculator/erlb

Note: Assume a blocking factor of 1% (0.010).

3 Find the TLAN bandwidth use (Kbps) in Table 36 on page 158 based on the codec used for the traffic source.

4 Calculate the bandwidth of a subnet using the following calculation:

Bandwidth for each subnet equals the total number of lines multiplied by the TLAN bandwidth usage, that is:

Subnet bandwidth = Total number of lines x TLAN bandwidth usage

5 Repeat steps 1 to 4 for each subnet.

6 To calculate the total TLAN traffic, add the total bandwidth for each subnet calculation.

End of Procedure

Table 36 on page 158 lists the bandwidth consumed by the various payload sizes of each codec. The Call Server uses the values in the columns labeled "TLAN Bandwidth". It looks up these values and subtracts them from the available zone bandwidth to determine if a zone has sufficient bandwidth for the call.

The "TLAN Bandwidth" values contain the total IP and Ethernet packet overhead of 78 bytes, including the 8 byte preamble and minimum 12 byte inter-packet gap. These are often excluded from bandwidth calculations but must be included to give a true indication of the bandwidth used. The Call Server assumes a Half Duplex Ethernet connection (again, to cover the worse case), so the bandwidth values shown are twice what is normally listed for a Full Duplex link.

The columns labeled "Base WAN Bandwidth" provide the data for the payload plus IP overhead without the Ethernet interface overhead. This data provides the basis for any WAN bandwidth calculations. The overhead associated with the particular WAN facility, such as Frame Relay, is added to the base value to determine the total bandwidth used. The values shown are for a duplex link, so if the WAN facility is Half Duplex, the values should be doubled.

> ***Note:*** The Call Server is unaware of the particulars of the WAN facility and always uses the values shown in the "TLAN Bandwidth" columns.

Table 36
TLAN and WAN IP bandwidth usage for each bi-directional IP conversation

				TLAN Bandwidth		Base WAN Bandwidth	
Codec type	Packet duration (ms)	Voice payload (bytes)	VAD	Peak bandwidth (Kbps)	Average bandwidth (Kbps)	Peak bandwidth (Kbps)	Average bandwidth (Kbps)
G.711 (64 Kbps)	10	80	Off	252.80	252.80	96.00	96.00
	20	160	Off	190.40	190.40	80.00	80.00
	30	240	Off	169.60	169.60	74.67	74.67
G.729A (8 Kbps)	10	10	Off	140.80	140.80	40.00	40.00
	20	20	Off	78.40	78.40	24.00	24.00
	30	30	Off	57.60	57.60	18.67	18.67
	40	40	Off	47.20	47.20	16.00	16.00
	50	50	Off	40.96	40.96	14.40	14.40
G.729AB (8 Kbps)	10	10	On	140.80	84.48	40.00	24.00
	20	20	On	78.40	47.04	24.00	14.40
	30	30	On	57.60	34.56	18.67	11.20
	40	40	On	47.20	28.32	16.00	9.60
	50	50	On	40.96	24.58	14.40	8.64
G.723.1 (6.3 Kbps)	30	24	Off	54.40	54.40	17.07	17.07
G.723.1 (5.3 Kbps)	30	24	Off	54.40	54.40	17.07	17.07

Note 1: The bandwidth estimates in Table 36 on page 158 assume a Half Duplex LAN/WAN connection and show the total bandwidth (both directions) used. For a Full Duplex link, such as Full Duplex Ethernet, the table's bandwidth estimates should be divided by two to determine the bandwidth used in one direction. However, the Call Server assumes the worse case (Half Duplex) when calculating the bandwidth used for each call against a Zone's bandwidth capacity.

Note 2: The "TLAN Bandwidth" overhead is assumed to be for Ethernet connections and is comprised of 8 bytes of Ethernet Preamble, 14 bytes of Ethernet, 20 bytes of IP, 8 bytes of UDP, and 12 bytes of RTP Header, plus 4 bytes of Ethernet check sum and 12 bytes of inter-packet gap. The total packet payload overhead is 78 bytes.

Note 3: Different transport types have slightly different bandwidth requirements.

Note 4: The average bandwidth is reduced from the peak bandwidth by the use of silence suppression (VAD).

Note 5: The reduction due to VAD is assumed to be 40%.

Note 6: The "Base WAN Bandwidth" overhead includes only the IP packet overhead of 20 bytes of IP, 8 bytes of UDP, and 12 bytes of RTP Header. The total packet payload overhead is 40 bytes. The values shown are for a Full Duplex data rate. Double the values if the WAN facility is Half Duplex.

TLAN engineering example

The following is an example of calculating TLAN bandwidth assuming Half Duplex links.

1 Subnet A: 28 Internet Telephones x 6 CCS/36 = number of Erlangs

Using G.729AB 30 msec, TLAN bandwidth usage is 57.6 Kbps.

Subnet A bandwidth = 4.66 lines x 57.6Kbps = 268.4 Kbps

2 Subnet B: 72 Internet Telephones, average 5 CCS/Internet Telephone

Subnet B total Erlangs = 72 x 5/36 = 10

Subnet B bandwidth = 10 x 57.6 = 576 Kbps

3 Subnet C: 12 Internet Telephones, average 6 CCS/Internet Telephone

Subnet C total Erlangs = 12 x 6/36 = 2

Subnet C bandwidth = 2 x 57.6 = 115.2 Kbps

4 Calculate the TLAN Bandwidth by adding each subnet bandwidth:

TLAN Bandwidth = 268.4 + 576 + 115.2 = 959.6 Kbps

End of Example

Assess WAN link resources

If IP Line traffic is routed over an intranet, the technician must assess the status of the network. For a locally connected Internet Telephone, if calls are routed to the PSTN, the calls affect only the capacity of the TLAN.

When calls are routed through an intranet, WAN links are frequently the source of capacity problems in the network. Unlike LAN bandwidth, which is virtually free and easily implemented, WAN links take time to obtain financial approval, provision, and upgrade. It is important to assess the state of WAN links in the intranet prior to implementing the IP Line network.

WAN traffic calculations

For data rate requirements for the intranet route, calculation is based on duplex channels. The data rate for a WAN is the duplex data rate. For example, 128 Kbps on the LAN is equal to a 64 Kbps duplex channel on the WAN. Use the following procedure to calculate data rate requirements for the intranet route. The effects of Real-time Transport Protocol (RTP) header compression by the router are not considered in these calculations but must be included where applicable.

Procedure 3
Calculating WAN traffic

1. Total subnet traffic = Number of Internet Telephones × CCS/Internet Telephone.
2. Convert to Erlangs:

   Total CCS / 36 (on the Half Duplex LAN)
3. Find WAN bandwidth usage (Kbps) from the "WAN Base Bandwidth" columns of Table 35 on page 154.
4. Bandwidth for each subnet = Total Erlangs × WAN bandwidth usage.
5. Multiply bandwidth of each subnet by 1.3 to adjust for traffic peaking.
6. Repeat the procedure for each subnet.

7 Adjust WAN bandwidth to account for WAN overhead depending on the WAN technology used:

- ATM (AAL1): multiply subnet bandwidth × 1.20 (9 bytes overhead/44 bytes payload)
- ATM (AAL5): multiply subnet bandwidth × 1.13 (6 bytes overhead/47 bytes payload)
- Frame Relay: multiply subnet bandwidth × 1.20 (6 bytes overhead/30 bytes payload – variable payload up to 4096 bytes)

Note: Each WAN link must be engineered to be no more than 80% of its total bandwidth if the bandwidth is 1536 Kbps or higher (T1 rate). If the rate is lower, up to 50% loading on the WAN is recommended.

End of Procedure

WAN engineering example

The following is an example of calculating the WAN bandwidth.

1 Subnet A: 36 Internet Telephones, average 6 CCS/Internet Telephone

- Total Erlangs = 36 x 6/36 = 6
- For G.729AB 50 msec, WAN bandwidth usage is 14.4 Kbps.
- Subnet A WAN bandwidth = 14.4 x 6 = 86.4Kbps
- Subnet A WAN bandwidth with 30% peaking
  = 86.4 x 1.3
  = 112.32 Kbps

2 Subnet B: 72 Internet Telephones, average 5 CCS/Internet Telephone

- Total Erlangs = 72 x 5/36 = 10
- Subnet B WAN bandwidth = 14.4 x 10 = 144 Kbps
- Subnet B WAN bandwidth with 30% peaking
  = 144 x 1.3
  = 187.2 Kbps

3 Subnet C: 12 Internet Telephones, average 6 CCS/Internet Telephone

- Total Erlangs = 12 x 6/36 = 2
- Subnet C WAN bandwidth = 14.43 x 2 = 28.8 Kbps
- Subnet C WAN bandwidth with 30% peaking
  = 28.8 x 1.3
  = 37.44 Kbps

4 If the WAN is known to be an ATM network (AAL1), the estimated bandwidth requirements are:

- Subnet A WAN bandwidth with ATM overhead
  = 112.32 × 1.2
  = 134.78 Kbps.
- Subnet B WAN bandwidth with ATM overhead
  = 187.2 × 1.2
  = 224.64 Kbps
- Subnet C WAN bandwidth with ATM overhead
  = 37.44 × 1.2
  = 44.93 Kbps

Note: Bandwidth values can vary slightly depending on the transport type.

End of Example

VoIP bandwidth management zones

Each Internet Telephone and Voice Gateway Media Card port is assigned a zone number in which it resides. The zone indicates the VoIP bandwidth management zone of the IP devices so that IP bandwidth can be managed within locations and between locations. This enables users to avoid quality degradation because of insufficient bandwidth for active connections.

For example, a branch office or telecommuter location can have more Internet Telephones than are supported by the IP link to that location (for example, 128 Kbps bandwidth with 10 Internet Telephones).

The zones are also used to determine if voice compression and silence detection is used for a connection.

Zone properties are defined in LD 117. Up to 256 zones can be configured. The Meridian 1 and Succession CSE 1000 systems use the zones for bandwidth management. New calls are blocked when the bandwidth limit is reached.

Each zone has four parameters. The prompt lists the parameters as p1, p2, p3, p4, and p5:

- p1 - The total bandwidth available for intrazone calls.
- p2 - The preferred strategy for the choice of codec for intrazone calls (that is, preserve best quality or best bandwidth).
- p3 - The total bandwidth available for interzone calls.
- p4 - The preferred strategy for the choice of the codec for interzone calls.
- p5 - The zone resource type; the type is either shared or private.

The Call Server uses the values shown in Table 36 on page 158 when calculating the bandwidth each call uses in a zone. The Call Server cannot determine whether the LAN/WAN connection is Half or Full Duplex. Therefore, the Call Server assumes the worse case, and subtracts the bandwidth consumed on a Half Duplex link by the codec and voice payload combination from the available zone bandwidth.

This should be considered when entering a zone's intra and inter bandwidth values in LD 117. If the zone has Full Duplex links, then the bandwidth entered should be doubled. For example, with a 100 BaseT Full Duplex LAN, the intra zone bandwidth can be configured to be 200000.

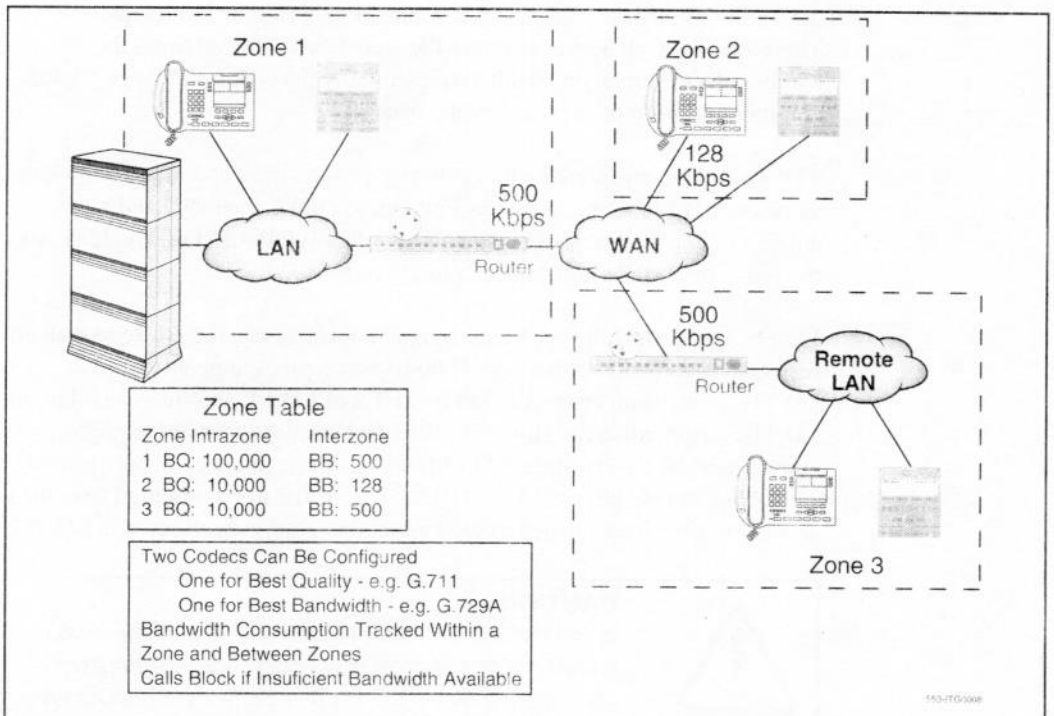
If no IP voice zones are configured, zone 0 operates as a default zone with no restrictions on bandwidth usage. If no IP voice zones are configured in LD 117, zone 0 can be configured for IPTN in LD 14, and for virtual line in LD 11 as a default zone. However, if any additional zones are required, zone 0 must be first configured in LD 117 if it is referenced by any Internet Telephone or ITG Physical TNs (IPTN). If zone 0 is not configured first, then all calls in zone 0 are labeled as soon as another zone is configured in LD 117.

CAUTION
When moving an Internet Telephone, the Administrator should check and change, if necessary, the telephone's zone assignment in LD 11. See *Software Input/Output: Administration* (553-3001-311).

CAUTION
Zone 0 must be configured in LD 117 before other zones are configured or all calls associated with zone 0 are blocked.

Figure 9 on page 166 shows an example of bandwidth management.

Figure 9
Bandwidth management example



Note: Bandwidth values given are for available bandwidth for VoIP and not for total bandwidth capacity.

Relationship between zones and subnets

Internet Telephones and Voice Gateway Media Cards gateway ports are assigned to zones based on the bandwidth management requirements of the particular installation. Devices in different subnets must traverse a router to communicate and can lie on different ends of a WAN facility. When Internet Telephones and gateway ports are in different subnets, the network facilities between them must be examined to see if it warrants placing the separated devices in different zones.

It is not necessary to always assign different zones. For instance, there can be different subnets within a LAN interconnected by router(s) with sufficient bandwidth. The Internet Telephones and gateway channels spread across them could all reside in a single zone. However, if there is a WAN facility with limited bandwidth between two subnets, the devices on the opposite ends should be placed in different zones so the bandwidth across the WAN can be managed.

Link utilization assessment

To assess the link utilization, follow the steps in Procedure 4.

Procedure 4
Assessing link utilization

1 Obtain a current topology map and link utilization report of the intranet.

2 Visually inspect the topology map to reveal which WAN links are likely to be used to deliver IP Line traffic. Alternately, use the traceroute tool (see "The following measuring tools are based on the ICMP (Internet Control Messaging Protocol):" on page 175).

3 Find out the current utilization of the WAN links. For example, the link's use can be averaged over a week, a day, or an hour.

4 Obtain the busy period (peak hour) use of the link.

5 Since WAN links are Full Duplex and data services exhibit asymmetric traffic behavior, obtain the utilization of the link representing traffic flowing in the heavier direction.

6 Assess how much spare capacity is available.

Enterprise intranets are subject to capacity planning policies that ensure that capacity usage remains below some pre-determined use level.

For example, a planning policy states that the use of a 56 Kbps link during the peak hour must not exceed 50%; for a T1 link, the threshold is higher, perhaps 80%. The carrying capacity of the 56 Kbps link would therefore be 28 Kbps, and for the T1, 1.2288 Mbps. In some organizations, the thresholds can be lower than that used in this example; in the event of link failures, there needs to be spare capacity for traffic to be re-routed.

7 The difference between the current capacity, and its allowable limit, is the available VoIP capacity.

For example, a T1 link used at 48% during the peak hour, with a planning limit of 80% has an available capacity of about 492 Kbps.

End of Procedure

Estimating network loading due to IP Line

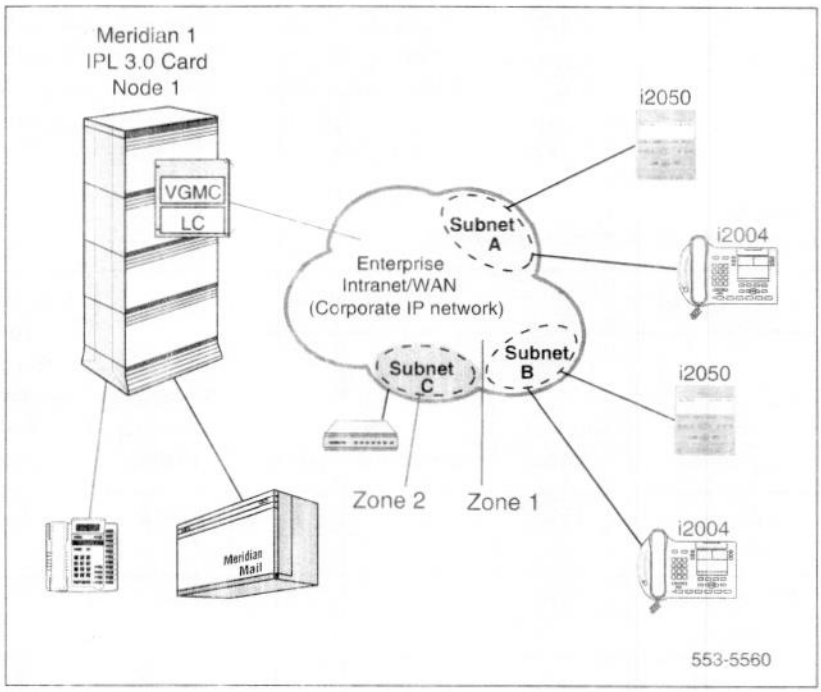
At this point, the technician has enough information to "load" the IP Line traffic on the intranet. The following example illustrates how this is done on an individual link. Not only must the IP Line traffic be taken into account but also the ITG Trunk traffic.

Example:

The intranet has a topology as shown in Figure 10 on page 169, and the technician wants to predict the amount of traffic between the IP Telephony node and corporate intranet. From the Calculate IP Line traffic requirements section (see page 155) and traceroute measurements, the traffic is collected between the IP Telephony node and subnet A, the IP Telephony node and subnet B, and the IP Telephony node and Router/Server C.

To complete this example, the traffic flow from the IP Telephony node to all routes needs to be totaled to determine the load to the link (TLAN).

Figure 10
An IP Line intranet with subnetworks



Decision: Is there sufficient capacity?

A link is defined as the route between the IP Telephony node and a subnet. Table 37 organizes the computations for each link, so that the available link capacity can be compared against the additional Voice Gateway Media Card load. For example, on the link from the IP Telephony Node to Subnet C, there is plenty of available capacity (568 Kbps) to accommodate the additional 24 Kbps of Voice Gateway Media Card traffic.

Table 37
Link Utilization Summary Example

Link		Utilization (%)		Available capacity (Kbps)	Incremental Voice Gateway Media Card Traffic (Kbps)	Sufficient capacity?
End-points	Capacity (Kbps)	Threshold	Used			
IPLine_Node1 - SubnetA	1536	80	75	76.8	72.5	Yes
IPLine_Node1 - SubnetB	1536	80	50	460.8	120.9	Yes
IPLine_Node1 - SubnetC	1536	80	48	492	24.2	Yes

Some network management systems have network planning modules that compute network flows in the manner just described. These modules provide detailed and accurate analysis as they take into account actual node, link, and routing information. They also help the technician assess the network resilience by conducting link and node failure analysis. By simulating failures, re-loading the network, and re-computing routes, the modules indicate where the network can run out of capacity during failures.

Insufficient link capacity

If there is insufficient link capacity, consider upgrading the link's bandwidth. RTP header compression can be implemented on the WAN router if it is a narrow bandwidth WAN link (less than 1.5 Mbps).

Set service parameters

Quality of Service (QoS) mechanism

The QoS requested from the IP network is controlled by the DiffServ Code Point (DSCP). QoS is controlled by setting the DSCP field in the IP header for both the Voice Gateway Media Card and the Internet Telephone. Individual values are configurable for the voice and control DSCP values and can be configured to a number between 0 and 63 inclusive. DSCP values control per hop behavior for packet forwarding for the router. The values are set once for each system and apply to all packets sent by the Voice Gateway Media Card and the Internet Telephone.

If DiffServ is implemented on the network, ask the network administrator for the default values that are used in the system. The recommended configuration values are:

- voice DSCP: 46 - Expedited Forwarding (EF) per hop behavior
- control DSCP: 40 - Class Selector 5 (CS5)

  Note: OTM and Element Management have 46 and 40 as the default values.

In some cases, the IP Network Administrator can set the DiffServ field at the edge of the QoS-controlled network by routers at the network edge. DiffServ can be set based on source or destination address, or port number. The Voice Gateway Media Card and Internet Telephone can be configured to have RTP voice UDP port numbers in a specific range.

Voice Gateway Media Card and Internet Telephone port numbers

Table 38 lists the UDP ports used for IP Line to Internet Telephone communications.

Table 38
UDP ports used for IP Line to Internet Telephone communications (Part 1 of 2)

Port usage	Port number mapping
Signaling (UNIStim over RUDP link)	IP Line port 5100 to Internet Telephone port 5000 ***Note:*** Port 5000 is fixed by the Internet Telephone.
Voice (Media)	RTP: • ITG-P Line Card IP Line port 5200 - 5246 (even numbers) to Internet Telephone port 5200 • Succession Media Card IP Line port 5200 - 5262 (even numbers) to Internet Telephone port 5200 ***Note:*** The OTM user interface enables setting a "Voice Port" value; this sets both the Internet Telephone RTP port and the starting port for the IP Line gateway's RTP port range. 5200 is the default. The Internet Telephone uses port 5200 while the IP Line gateway channels [0-23] use the even port numbers in the range [5200 - 5246]. RTCP: • ITG-P Line Card IP Line port 5201 - 5247 (odd numbers) to Internet Telephone port 5201 • Succession Media Card IP Line port 5201 - 5263 (odd numbers) to Internet Telephone port 5201 ***Note:*** The RTCP port numbering is based on each channel's RTP port + 1. Therefore, the RTCP port range is dependent on the configuration for RTP port.

Table 38
UDP ports used for IP Line to Internet Telephone communications (Part 2 of 2)

Port usage	Port number mapping
Registration	IP Line ports 4100 and 7300 to Internet Telephone port 5000
i2002/i2004 Internet Telephone firmware download	TFTP standard port 69

Table 39 lists the other UDP ports used by the IP Line.

Table 39
Other UDP ports used by the IP Line

Port usage	Port number mapping
Bidirectional TPS to TPS signaling (intercard)	16543
SNTP signaling	20000 + IP Telephony node number
BOOTP server on Leader	67 (standard)
BOOTP client on Followers	68 (standard)
SNMP on ELAN interface	161 (standard)
RUDP signaling with Meridian 1 core CPU on ELAN interface	15000 15001

Measure intranet Quality of Service

Utilization of the existing data network must be assessed to determine the quality of voice services it can support.

End-to-end delay and error characteristics of the intranet must be measured so that the technician can set realistic QoS expectations for intranet voice services.

WARNING
Network designers must be aware of traffic calling patterns between any combination of Internet Telephones and gateway channels, and must plan the capacity of connecting elements to handle the expected traffic.

The use of measuring tools requires a source node and a destination node. The source node can be a "PING" (see page 175) host on a LAN segment attached to the router intended to support the node. The destination node can be a remote subnet. The requirement is briefly described as follows.

Note: Ensure that the ITG network DiffServ bytes are set to their intended operational values before taking measurements.

Criteria

- **End-to-end packet delay:** Packet delay is the point-to-point, one-way delay between the time a packet is sent to the time it is received at the remote end. It is comprised of delays at the Voice Gateway Media Card, Internet Telephone, and the IP network. To minimize delays, the IP Telephony node and Internet Telephone must be located to minimize the number of hops to the network backbone or WAN.

  Note: To ensure good voice quality, an end-to-end delay of <= 50 ms is recommended on the IP network. This does not include the built-in delay of the Voice Gateway Media Card and Internet Telephone.

- **End-to-end packet loss:** Packet loss is the percentage of packets sent that do not arrive at their destination. Transmission equipment problems, packet delay, and network congestion cause packet loss. In voice conversation, packet loss appears as gaps in the conversation. Sporadic loss of a few packets can be more tolerable than infrequent loss of a large number of packets clustered together.

  Note: For high-quality voice transmission, the long-term average packet loss between the Internet Telephones and the Voice Gateway Media Card TLAN interface must be < 1%, and the short-term packet loss must not exceed 5% in any 10-second interval.

  Packet loss on the ELAN interface can cause:

  — communication problems between the Call Server and the Voice Gateway Media Cards

  — lost SNMP alarms

  — incorrect status information on the OTM console

  — other signaling related problems

  Note: Since the ELAN network is a Layer 2 Switched LAN, the packet loss must be zero. If packet loss is experienced, its source must be investigated and eliminated. For reliable signaling communication on the ELAN interface, the packet loss must be < 1%.

The following measuring tools are based on the ICMP (Internet Control Messaging Protocol):

- PING (sends ICMP echo requests)
- Traceroute (sends packets to unequipped port numbers and processes to create ICMP destination unavailable messages).

Both PING and traceroute are basic measuring tools that can be used to assess the IP Line network. They are standard utilities that come with most commercial operating systems. PING is used to measure the round-trip delay of a packet and the percentage of packet loss. Traceroute breaks down delay segments of a source-destination pair and any hops in-between to accumulate measurements.

There are several third-party applications that perform data collection similar to PING and traceroute. In addition, these programs analyze data and plot performance charts. The use of PING and traceroute to collect data for manual analysis is labor intensive; however, they provide information as useful as the more sophisticated applications.

The following analysis use PING/traceroute data for discussion, although it is likely in most situations a third-party application is used.

Destination Types

To a remote subnet

This configuration involves an intranet subnet that is attached to a number of Internet Telephones, that becomes an intermediate hop in the path delivering voice packets between the Internet Telephone and the IP Line network. Collect the delay measurement between the PING host and the subnet server.

Measuring end-to-end network delay

The basic tool used in IP Line networks to measure end-to-end network delay is the PING program. PING takes a delay sample by sending an ICMP packet from the host of the PING program to a destination server, and waits for the packet to make a round trip.

To ensure the delay sample results are representative of the IPLine_Node1:

a Attach the PING host to a "healthy" LAN segment.

b Attach the LAN segment to the router intended to support the IP Telephony node.

c Choose a destination host by following the same critical guidelines as for the source host.

The size of the PING packets can be any number; the default is 60 bytes.

Sample PING output:

```
IPLine_Node1% PING -s subnetA 60
PING subnetA (10.3.2.7): 60 data bytes
68 bytes from (10.3.2.7): icmp_seq=0 ttl=225 time=97ms
68 bytes from (10.3.2.7): icmp_seq=0 ttl=225 time=100ms
68 bytes from (10.3.2.7): icmp_seq=0 ttl=225 time=102ms
68 bytes from (10.3.2.7): icmp_seq=0 ttl=225 time=97ms
68 bytes from (10.3.2.7): icmp_seq=0 ttl=225 time=95ms
68 bytes from (10.3.2.7): icmp_seq=0 ttl=225 time=94ms
68 bytes from (10.3.2.7): icmp_seq=0 ttl=225 time=112ms
68 bytes from (10.3.2.7): icmp_seq=0 ttl=225 time=97ms
^?
--- IPLine_Node1 PING Statistics ---
8 packets transmitted, 8 packets received, 0% packet loss
round-trip (ms) min/avg/max = 94/96/112
```

Note: PING results can vary.

Assessment of sample PING output

Note: The round-trip time (rtt) is indicated by the time field.

The rtt from the PING output varies. It is from repeated sampling of rtt that a delay characteristic of the intranet can be obtained. To obtain a delay distribution, the PING tool can be embedded in a script that controls the frequency of the PING probes, timestamps and stores the samples in a raw data file. The file can then be analyzed later using a spreadsheet or another application. The technician can also check if the intranet's network management software has any delay measurement modules that can obtain a delay distribution for a specific route.

Delay characteristics vary depending on the site pair and the time-of-day. The site pair is defined as the measurement between the host IP Line and the remote subnet (for example, IP Line to subnet A in Figure 10 on page 169). The assessment of the intranet must include taking delay measurements for each IP Line site pair. If there is a significant variation of traffic on the intranet, include PING samples during the intranet's peak hour. For a complete assessment of the intranet's delay characteristics, obtain PING measurements over a period of at least a week.

Measuring end-to-end packet loss

The PING program also reports whether the ICMP packet successfully completed its round trip. Use the same PING host setup to measure end-to-end error, and in making delay measurement, use the same packet size parameter.

Multiple PING samples must be used when sampling for error rate. Packet loss rate (PLR) is the error rate statistic collected by multiple PING samples. To be statistically significant, at least 300 samples must be used. Obtaining an error distribution requires running PING over a greater period of time.

Recording routes

The traceroute tool records routing information for all source-destination pairs as part of the network assessment. An example of the traceroute output is shown below:

```
ipline_node1% traceroute subnetA
traceroute to subnetA 10.3.2.7, 30 hops max, 32 byte packets
1   r6 (10.8.0.1) 1 ms  1 ms  1 ms
2   r5 (10.18.0.2) 42 ms  44 ms  38 ms
3   r4 (10.28.0.3) 78 ms  70 ms  81 ms
4   r1 (10.3.0.1) 92 ms  90 ms  101 ms
5   subnetA (10.3.2.7) 94 ms  97 ms  95 ms
```

The traceroute program is also used to verify whether routing in the intranet is symmetric or not for each of the source-destination pairs. This is done using

the -g loose source routing option, as illustrated in the following command syntax:

```
ipline_node1% traceroute -g subnetA ipline_node1
```

Adjusting PING measurements

One-way and roundtrip

The PING statistics are based on round-trip measurements, while the QoS metrics in the Transmission Rating model are one-way. Divide the delay and packet error PING statistics in half to ensure the comparison is valid.

Adjustment due to IP Line processing

The PING measurements are taken from PING host to PING host. The Transmission Rating QoS metrics are from end-user to end-user, and include components outside the intranet. The PING statistic for delay needs to be further modified by adding 93ms to account for the processing and jitter buffer delay of the nodes.

Note: No adjustment needs to be made for error rates.

If the intranet measurement barely meets the round-trip QoS objectives, the technician must be aware of the possibility that the one-way QoS is not being met in one of the directions of flow. This can apply even if the flow is on a symmetric route due to asymmetric behavior of data processing services.

Late packets

Packets that arrive outside the window allowed by the jitter buffer are discarded by the IP Line. To determine which PING samples to ignore, calculate the average one-way delay based on all the samples.

To calculate late packets, double the value of the nominal jitter buffer setting. For example, assume:

- the average one-way delay is 50 msec
- the jitter buffer is set to a nominal (or average) value of 40 msec
- then the maximum value is 2 x 40 + 50 = 130 msec

Therefore, any packet with a one-way delay of greater than 130 msec is late, and must be added to the total number of packets lost.

Estimate Voice Quality

The perceived quality of a telephone call is dependent on many factors, such as codec characteristics, end-to-end delay, packet loss, and the perception of the individual listener.

The E-Model Transmission Planning Tool is a model used to produce a quantifiable measure of voice quality based on relevant factors. Refer to two ITU-T recommendations (ITU-T E.107 and E.108) for more information on the E-Model and its application.

A simplified version of the E-Model is applied to the Internet Telephone to provide an estimate of the voice quality that the user can expect based on various configuration choices and network performance metrics.

The simplified E-Model is given below:

R = 94 - Ic - Id - Ip

where:
Ic = codec impairment (see Table 40 on page 181)

Id = delay impairment (see Table 41 on page 181)

Ip = packet loss impairment (see Table 42 on page 182)

Note: This model already takes into account some characteristics of the Internet Telephone, and therefore the impairment factors are not identical to those shown in the ITU-T standards.

Refer to Table 43 on page 182 for the translation of R values into user satisfaction levels.

Table 40
Impairment factors of codecs

Codec	Codec Impairment (Ic) (msec frames)
G.711	0
G.729A/AB	11 - 20 or 30
G.729A/AB	16 - 40 or 50
G.723.1 (5.3 Kbps)	19
G.723.1 (6.3 Kbps)	15

Table 41
Impairment factors due to network delay

Network delay* (msec)	Delay Impairment (Id)
0 -49	0
50 - 99	5
100 -149	10
150 - 199	15
200 - 249	20
250 - 299	25
* Network delay is the average one-way network delay plus jitter.	

Table 42
Impairment factors due to packet loss

Packet loss (%)	Packet Lose Impairment (Ip)
0	0
1	4
2	8
4	15
8	25

Table 43
R value translation

R Value (lower limit)	MOS	User Satisfaction
90	4.5	Very satisfied
80	4.0	Satisfied
70	3.5	Some users dissatisfied
60	3.0	Many users dissatisfied
50	2.5	Nearly all users dissatisfied
0	1	Not recommended

Sample scenarios

Scenario 1

A local LAN has the following characteristics:

- G.711 codec
- 20 msec network delay
- 0.5% packet loss

To calculate R = 94 - Ic - Id - Ip, use Table 40, Table 41, and Table 42:

- G.711 codec: Ic = 0
- 20 msec network delay: Id = 0
- 0.5% packet loss: Ip = 2

Then:

R = 94 - 0 - 0 - 2

R = 92

Using Table 43 on page 182, a value of 92 means the users are very satisfied.

Scenario 2

A campus network has the following characteristics:

- G.711 codec
- 50 msecs delay
- 1.0% packet loss

To calculate R = 94 - Ic - Id - Ip, use Table 40, Table 41, and Table 42:

- G.711 codec: Ic = 0
- 20 msec network delay: Id = 5
- 0.5% packet loss: Ip = 4

Then:

R = 94 - 0 - 5 - 4

R = 85

Using Table 43 on page 182, a value of 85 means the users are satisfied.

Scenario 3

A WAN has the following characteristics:

- G.729 codec
- 30 msec network delay
- 2% packet loss.

To calculate R = 94 - Ie - Id - Ip, use Table 40, Table 41, and Table 42:

- G.711 codec: Ie = 11
- 20 msec network delay: Id = 5
- 0.5% packet loss: Ip = 8

Then:

R = 94 - 11 - 5 - 8

R = 70

Using Table 43 on page 182, a value of 70 means some users are dissatisfied.

DiffServ

The Differentiated Service (DiffServ) determines the priority of the packets in the IP network. The value entered depends on the equipment in the data network. The DiffServ applies to all cards in the node and also applies to any Internet Telephones that register with this node.

Individual values are configurable for the voice and control DiffServ Code Point (DSCP) values and can be configured to a number between 0 and 63 inclusive using OTM 2.0.

The values are set once for each system and apply to all packets sent by the Voice Gateway Media Card and the Internet Telephone.

If DiffServ is implemented on the network, ask the network administrator for the default values that are used in the system. The recommended configuration values are:

- voice DSCP: 46 - Expedited Forwarding (EF)
- control DSCP: 40 - Class Selector 5 (CS5)

Note: OTM and Element Management has 46 and 40 as the default values.

Loss and Level Plan

The Voice Gateway Media Card ships with a predefined loss and level plan. The loss and level plan determines various parameters, such as transmission gain, that vary from country to country. The default loss and level plan is for the United States. The values are stored in a file on the OTM PC. You can select other countries when you configure the DSP Profile settings in OTM's ITG Line 3.0 application or the Loss Plan settings in Element Management.

Note: A dynamic loss plan has been implemented on the Succession CSE 000 Rel 2.0 system. It uses PDCA Table 15 in LD 73 to define the gateway loss value per endpoint connection type and adjusts the Voice Gateway Media Card gateway channel's loss for each call by sending pad values to the card. The default values in the system are for the North American loss plan. Installation of the Succession CSE 000 Rel 2.0

system in any other country requires setting the pad values in Table 15 to that country's loss plan. See the section on Transmission Parameters in Planning and Engineering Guidelines (553-3023-102) for details.

Echo canceller

OTM supports echo canceller tail lengths of 8, 16, and 32 msec. The default in OTM is the maximum of 32 msec. Element Management supports echo canceller tail lengths of 0, 64, and 128 (default) msec. It is recommended that the maximum echo canceller tail length is used. Never reduce the echo canceller tail delay value unless directed by Nortel Networks Field Support.

OTM's scaling process

The IP Line loadware contains an enhanced echo canceller. The new echo canceller has improved echo cancellation algorithms and supports tail lengths up to 128 msec. OTM 1.0.15 does not offer the enhanced tail length values in the echo canceller tail pull down menu. Because the IP Line loadware can be used with any supported version of the OTM product, it scales the configured tail length, if necessary, so that the maximum tail length is achieved. The scaling provides the following echo cancelling tail lengths:

8 -> 32 msec
16 -> 64 msec
32 -> 128 msec

The maximum tail length is the recommended value. Selecting the OTM default of 32 msec yields the desired maximum tail length of 128 msec.

An update to OTM changes the echo canceller tail configuration pull down list to the values 64 and 128. When used with the IP Line loadware, the application no longer needs to scale the configured value. The configured value matches the tail length used by the echo canceller.

Note: If the OTM version 2.0 is used with an ITG Line 2.0 loadware version, such as ITG Line 2.01.53, selecting 64 or 128 msecs sets the echo tail length to 32 msec.

Reducing delays

The link delay is the time it takes for a voice packet to be queued on the transmission buffer of a link until it is received at the next hop router. Link delay can be reduced by:

- Upgrading link capacity. This reduces the serialization delay of the packet, but also reduces the utilization of the link and the queueing delay. Before upgrading a link, the technician must check both routers connected to the link to be upgraded and ensure compliance with router configuration guidelines.
- Implementing a priority queueing discipline.

To determine the links for upgrading, list all the intranet links used to support the IP Line traffic. This can be derived from the traceroute output for each site pair. Use the intranet link utilization report and note the highest used links and the slowest links. Estimate the link delay of suspect links using the traceroute results.

Example: A 256 Kbps link from router1 to router 2 has a high utilization. The following is a traceroute output that traverses this link:

```
IPLine_Node1% traceroute SubnetA
traceroute to SubnetA (10.3.2.7), 30 hops max, 32 byte packets
    router1 (10.8.0.1) 1 ms 1 ms  1 ms
    router2 (10.18.0.2) 42 ms  44 ms  38 ms
    router3 (10.28.0.3) 78 ms  70 ms  81 ms
    router4 (10.3.0.1) 92 ms  90 ms  101 ms
    SubnetA (10.3.2.7) 94 ms  97 ms  95 ms
```

The average rtt time on the example link is about 40 ms; the one-way link delay is about 20 ms, of which the circuit transmission and serialization delay are just a few milliseconds. Most of this link's delay is due to queueing.

Reducing hop count

The IP Telephony nodes must be connected to the intranet to minimize the number of router hops between the Voice Gateway Media Card and the Internet Telephone. This reduces the fixed and variable IP packet delay, and improves the Voice over IP Quality of Service. It is recommended that no more than one card use a particular 10BaseT LAN collision domain.

> ***Note:*** In a passive Ethernet hub, all ports on the hub share one 10Mbps collision domain; in a switched Ethernet hub, each port has its own collision domain. Hubs should not be used anywhere in the network path between the internet telephone and VGMC TLAN interface. A switched network is recommended

The IP Telephony node and the TLAN router should be placed as close to the WAN backbone as possible to:

- Minimize the number of router hops.
- Segregate constant bit-rate VoIP traffic from bursty LAN traffic.
- Simplify the end-to-end QoS engineering for packet delay, jitter, and packet loss.

If an access router separates the IP Telephony node from the WAN router, there must be a high-speed link (for example, Fast Ethernet, FDDI, SONET, OC-3c, ATM STS-3c) between the access router and the WAN backbone router.

Reducing packet errors

Packet errors in intranets are generally correlated with congestion somewhere in the network. Bottleneck links occur where the packet errors are high because packets get dropped when they arrive faster than the link can transmit them. When highly used links are upgraded, the sources of packet errors on a particular flow must be removed. A reduction in hop count also reduces the opportunities for routers and links to drop packets.

Other causes of packet errors, not related to queueing delay:

- **Poor link quality**–the underlying circuit has transmission problems, high line error rates, or subject to frequent outages. The circuit is provisioned on top of other services, such as X.25, frame relay, or ATM. Check with the service provider for resolution.
- **Overloaded CPU**–this is another commonly-monitored statistic collected by network management systems. If a router is overloaded, it means that the router is constantly performing processing-intensive tasks, which impedes the router from forwarding packets. Find out what the threshold CPU utilization level is, and check if any suspect router conforms to the threshold. The router has to be re-configured or upgraded.
- **Saturation**–routers can also be overworked when there are too many high capacity and high traffic links configured on it. Ensure that routers are dimensioned according to vendor guidelines.
- **LAN saturation**–packets are dropped on under-engineered or faulty LAN segments.
- **Jitter buffer too small**–packets that arrive at the destination ITG, but are too late to be placed in the jitter buffer, are essentially loss packets.

Adjusting jitter buffer size

The jitter buffer parameters directly affect the end-to-end delay. Lowering the voice playout settings decreases one-way delay, but there is less waiting time for voice packets that arrive late.

The jitter buffer setting is configured on the voice gateway channels of the Voice Gateway Media Card and are sent out to the Internet Telephones. The jitter buffer size is set when you configure the DSP Profiles:

- in the OTM ITG Line 3.0 application (see Figure 40 on page 285), or
- in the selected codec in Element Management (see Figure 66 on page 349)

The jitter buffer is statically configured and is the same for all devices in the network. The jitter buffer size range is 0-200 milliseconds. The default jitter buffer value is 50 milliseconds. However, the jitter buffer setting that is used on the Voice Gateway Media Card is a multiple of the codec frame size. The setting is automatically adjusted to be greater than or equal to the jitter buffer value set in the DSP Profile tab. As each call is set up, the jitter buffer for each device is set to the nearest whole number increment of the selected codec frame size.

For example, if the jitter buffer is configured as the default 50 msec in the DSP Profiles, but a 20 msec codec is used, the jitter buffer is set to 60 msec, which is the nearest whole number increment.

50 msec / 20 msec = 2.5
2.5 rounded up to the nearest whole number increment is 3.
3 x 20 msec = 60 msec

If the jitter buffer is configured as zero, the depth of the jitter buffer is set to the smallest value the device can support. In practice, the optimum depth of the jitter queue is different for each call. For telephones that are on a local LAN connection, a short jitter queue is desirable to minimize delay. For telephones that are several router hops away, a longer jitter queue is required.

Lowering the jitter buffer size decreases the one-way delay of voice packets. If the setting for the jitter buffer size is too small, packets are discarded unnecessarily. Discarded packets result in poorer speech quality and can be heard as clicks or choppy speech.

If the technician decides to discard packets, to downsize the jitter buffer, the technician must do the following:

- **Check the delay variation statistics.**

  Obtain the one-way delay distributions originating from all source IP Line sites.

- **Compute the standard deviation of one-way delay for every flow.**

  Some traffic sources with few hop counts yield small delay variations, but it is the flows that produce great delay variations that should be used to determine whether it is acceptable to resize the jitter buffer.

- **Compute the standard deviation (σ) of one-way delay for that flow.**

  Do not set the set the jitter buffer size smaller than 2s.

Codec selection

To ensure optimal voice quality, minimize the number of compression and decompression stages and wherever bandwidth permits, use G.711 codec.

There is a potential to degrade the voice quality if codecs are cascaded. This can occur when there are multiple compression and decompression stages on a voice call. The more IP links used in a call, the more delay is added, and the greater the impact on the voice quality.

The following is a list of applications and devices that can impact voice quality, if you use a compression codec such as G.729A:

- Voice mail, such as Nortel Networks CallPilot, introduces another stage of compression and decompression.
- Conferences can double the number of IP links.
- ITG Trunks can add additional stages of compression and decompression.

Post-installation network measurements

The design process is continual, even after implementation of the ITG network and commissioning of voice services over the network. Network changes – in ITG traffic, general intranet traffic patterns, network policies, network topology, user expectations, and networking technology – can render a design obsolete or non-compliant with QoS objectives. The design needs to be reviewed periodically against prevailing network conditions and traffic patterns.

It is assumed that the customer's organization already has processes to monitor, analyze, and re-design both the Meridian 1 and Succession CSE 1000 network and the corporate intranet to maintain internal QoS standards. When operating an ITG network, additional processes must be developed to:

- collect, analyze, and forecast ITG traffic patterns

- monitor Operational Measurements (see below)
- implement changes in the ITG and intranet when planning thresholds are reached

By establishing these new processes, the ITG network can be managed to ensure that desired QoS objectives are met.

ITG Operational Measurement (OM)

The Voice Gateway Media Card collects Operational Measurements from the Internet Telephones and DSP channels and saves the information to a log file every 60 minutes. The Operational Measurements include:

- Internet Telephone Registration Attempted Count
- Internet Telephone Registration Confirmed Count
- Internet Telephone Unregistration Count
- Internet Telephone Audio Stream Set Up Count
- Internet Telephone Average Jitter (msec)
- Internet Telephone Maximum Jitter (msec)
- Internet Telephone Packets Lost/Late (%)
- Internet Telephone Total Voice Time (minutes and seconds)
- Gateway Channel Audio Stream Set Up Count
- Gateway Channel Average Jitter (msec)
- Gateway Channel Maximum Jitter (msec)
- Gateway Channel Packets Lost/Late (%)
- Gateway Channel Total Voice Time (minutes and seconds)

OM Report description

The OM log file is a comma-separated (.csv) file stored on the OTM server. Using OTM you can run an adhoc report or schedule a regular report. A new file is created for each month of the year in which OM data is collected. It can be read directly or imported to a spreadsheet application for post-processing and report generation. Collect these OM reports and store them for analysis. At the end of each month, identify the hours with the highest packet lost/late statistics and standard deviation statistics generated. Compare the data to target network QoS objectives.

Declines in QoS can be observed through the comparison of QoS between last period and current period. A consistent inferior measurement of QoS compared with the objective triggers an alarm. The customer must take steps to strengthen the performance of the route.The card creates a new log file each day. Files are automatically deleted after seven days.

IP Line ELAN and TLAN configuration

A subnet is defined as a remote network serving a collection of Internet Telephones, which is represented by a server or router communicating with the ITG processor for VoIP service (see Figure 10 on page 169).

General requirements

The general requirements are as follows:

- no foreign broadcast coming from other subnets
- no BootP relay agent requirement (only on ELAN router interface)
- no Network Address Translation (NAT) between Internet Telephone and IP Telephony node
- the TLAN cable between the ITG-P Line Card and the Layer 2 switch must be 50 meters or less
- disable spanning tree on the Layer 2 switch ports connected to the ELAN and TLAN ports of the Meridian 1 and Succession CSE 1000 components

Separate subnet configuration

Each Voice Gateway Media Card has two Ethernet ports, one for the Telephony LAN (TLAN) and one for the Embedded LAN (ELAN). The advantages of this configuration are:

- optimization of VoIP performance on the LAN segment by segregating it from ELAN traffic and connecting the TLAN as close as possible to the WAN router.
- making the amount of traffic on the TLAN more predictable for QoS engineering.
- enhanced network access security when the ELAN and customer's enterprise network are separate or connected through a fire-wall router:

— enables placement of the modem router on the isolated ELAN, protecting the customer's enterprise network from unauthorized modem router accesses

— protects the Meridian 1/Succession CSE 1000 ELAN from unauthorized access from the customer's enterprise network

CAUTION
Due to backwards compatibility, the user interface permits you to choose whether to use separate subnets. It is, however, mandatory to use separate subnets.

ELAN/TLAN Half and Full Duplex operation

The ELAN on the Voice Gateway Media Card operates at Half Duplex only and is limited to 10BaseT operation due to filtering on the Meridian 1 Option 11C back planes.

The TLAN on Voice Gateway Media Card operates at Half Duplex or Full Duplex and can run at 10BaseT or 100BaseT.

It is recommended that any network equipment connected to the ELAN or TLAN be set to Auto Sense / Auto Negotiate for correct operation. Although Full Duplex is preferred, it is not required. For example, for the IP Line application, Half Duplex has ample bandwidth for a Voice Gateway Media Card even with 24 busy channels, VAD disabled, and G.711 codec with 10ms voice range.

Mismatches can occur if devices are hard configured for speed and duplex mode. Every device and port must be correctly configured to avoid duplex mismatch problems that typically exhibit as lost packets and CRC errors. The Voice Gateway Media Card cannot be set for 100BaseT/Full Duplex operation, and as a result the card's TLAN operates in Auto Negotiate mode. Duplex mismatches and lost packets occur if the TLAN interface is not configured properly.

CAUTION

Duplex mismatches occur in the LAN environment when one side is set to Auto Negotiate and the other is hard configured.

The Auto Negotiate side adapts only to the speed setting of the fixed side. For duplex operations, the Auto Negotiate side sets itself to Half Duplex mode. If the forced side is Full Duplex, a duplex mismatch occurs.

Subnet configuration for TLAN and ELAN ports

Single subnet configuration implies the configuration and use of just one Ethernet interface, namely the ELAN interface, over which all voice and management traffic is routed. Single subnet configuration can also mean configuring both the TLAN and ELAN interfaces to be in the same subnet. Neither configuration is supported. The configuration of the ELAN and TLAN must be done such that both interfaces are used and the assigned IP addresses are in different subnets.

Separate or dual subnet configuration implies the configuration of both the TLAN and ELAN interfaces. All management traffic is routed over the ELAN, while all telephony traffic is routed over the TLAN. The ELAN connection is to a 10BaseT hub or switch, while the TLAN is connected to a 10/100BaseT switch.

For dual subnet configuration, the TLAN and ELAN subnets must not overlap. For example, the following configuration is not valid, as the TLAN and ELAN subnets overlap.

ELAN IP	10.0.0.136
ELAN Gateway	10.0.0.129
ELAN Subnet Mask	255.255.255.224
TLAN Node IP	10.0.0.56
TLAN Card IP	10.0.0.57
TLAN Gateway	10.0.0.1
TLAN Subnet Mask	255.255.255.0

The ELAN range of addresses, 10.0.0.129 to10.0.0.160, overlaps with the TLAN range of addresses, 10.0.0.1 to 10.0.0.255. This fails to keep with IP addressing practices, as it is equally valid to route to IP packets over either the TLAN or ELAN interface and the resulting behavior from such a setup is undetermined.

A better way to split these IP addresses is:

ELAN IP	10.0.0.136
ELAN Gateway	10.0.0.129
ELAN Subnet Mask	255.255.255.224
TLAN Node IP	10.0.0.56
TLAN Card IP	10.0.0.57
TLAN Gateway	10.0.0.1
TLAN Subnet Mask	255.255.255.128

In this example:

- TLAN IP addresses are in the range of 10.0.0.1 to 10.0.0.127
- ELAN IP addresses are in a separate subnet of 10.0.0129 to 10.0.0.160

This satisfies the IP addressing practice of engineering the network such that subnets do not overlap. However, this results in a smaller subnet for the TLAN address.

Spanning Tree Option on Layer 2 switches

Nortel Networks recommends disabling the spanning tree option on the Layer 2 switch ports that connect to the Meridian 1 and Succession CSE 1000 system's TLAN and ELAN interfaces.

This option is "enabled" by default on most Layer 2 switches. If the option is left enabled, the subsequent spanning tree discovery algorithm initiated when a device connected to a port is reset, rebooted, or unplugged/plugged-in, can interfere with the Master Election Process of the Meridian 1 and Succession CSE 1000 system devices. In most cases the Master Election Process recovers from this after a slight delay. However, to reduce the potential of unforeseen complications in this scenario it is recommended that the spanning tree option on these ports be disabled.

LAN Device Interface Names

The devices in the system have different Device Interface Names depending on whether is it is on the TLAN or on the ELAN. Table 44 on page 198 shows the Device Interface Name for the Succession Media Card, ITG-P Line Card, and the Signaling Server.

Table 44
LAN Device Interface Names

Device Type	TLAN/ELAN	LAN Device Interface Name
Succession Media Card	ELAN	ixpMac1
	TLAN	ixpMac0
ITG-P Line Card	ELAN	lnIsa0
	TLAN	lnPci1
Signaling Server	ELAN	fei0
	TLAN	fei1

Installation and Configuration Summary

Contents

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Internet Telephone configuration data summary sheet 210

Reference list

The following are references for this section:

- *Internet Terminals Description* (553-3001-217)
- *Software Input/Output: Maintenance* (553-3001-511)

Overview

This chapter provides a summary of the procedures required to install a new Meridian 1 and Succession Communication Server for Enterprise (CSE) 1000 IP Telephony node, add cards to the node, install the cards, transmit data to the cards, and install the Internet Telephones. It also includes information on what you need before beginning the installation procedures.

Be sure to read the "IP Network Engineering Guidelines" on page 133 before installing an IP Telephony node.

The follow chapter is divided into two summary sections.

- "Meridian 1 Rel 25.30 or Succession CSE 1000 Rel 1.1 systems" on page 200
- "Succession CSE 1000 Rel 2.0 systems" on page 204

Meridian 1 Rel 25.30 or Succession CSE 1000 Rel 1.1 systems

Before you begin:

1 Ensure that the system meets the following minimum requirements: Meridian 1 Release 25.30 system or Succession CSE 1000 Release 1.1 system.

2 Upgrade the Meridian 1 or Succession CSE 1000 keycode to expand the ISM system limit to support the number of Internet Telephones you plan to install. Refer to *Software Input/Output: Maintenance* (553-3001-511) to use:

   a. LD 22 command SLT to determine your Internet Telephone limit on the system.

   b. LD 143 to expand the ISM parameters (if required).

3 Verify that you have the Optivity Telephony Manager (OTM) version 2.0.

   a. Check the Nortel Networks Software Downloads Web site to determine the latest software version and required patches (PEPs).

Refer to Procedure 111, "Downloading files from the Nortel Networks Web site" on page 719.

b. Upgrade to the latest OTM version if necessary.

4 Create site name, system name, and customer number in the OTM Navigator. Specify the correct Meridian 1 and Succession CSE 1000 system type in order for OTM ITG IP Line 3.0 application to prompt for correct TN format.

5 Provision the IP network to support IP Telephony node and Internet Telephones.

a. Choose the Internet Telephone DHCP mode: Full, Partial, or None (static IP address). For Full DHCP mode, refer to "Configuring the DHCP server to support Full DHCP mode" on page 215.

b. Determine the Internet Telephone 10/100BaseT Ethernet LAN connection: desktop switch, or a separate cable to the equipment closet.

c. Determine TLAN, ELAN IP address and Ethernet connections from the network IP administrator. Refer to the "IP Network Engineering Guidelines" on page 133.

6 Check that the required LAN and WAN networking equipment and cables are installed. For IP networking requirements, refer to "IP Network Engineering Guidelines" on page 133.

Installation summary for Meridian 1 Rel 25.30 and Succession CSE 1000 Rel 1.1 systems

The following summary of steps can be used as a reference guide to install and configure an IP Telephony node and Voice Gateway Media Cards on Meridian 1 and Succession CSE 1000 Release 1.1 systems. This summary is intended to serve as a pointer to the more detailed procedures contained in other chapters and to provide a sequential flow to the steps involved in the overall installation procedure.

Note: Complete all installation and configuration steps before you transmit data to the Voice Gateway Media Cards.

1 Complete the Voice Gateway Media Card installation summary sheet. Refer to Table 45 on page 209.

2 Complete the Internet Telephone configuration data summary sheet. Refer to Table 46 on page 210.

3 Install the hardware components:

   a. Install and cable the Voice Gateway Media Card(s). See Procedure 5 on page 230 for installing the ITG-P Line Cards and to Procedure 7 on page 236 for installing the Succession Media Card.

   b. Install an ITG Line-specific I/O Panel Filter Connector (Option 51C/61C/81/81C only). See Procedure 8 on page 238.

   c. Cable the Voice Gateway Media Card:

      i. Install the ELAN, TLAN, serial interface cable for the ITG-P Line Card. See Procedure 9 on page 243.

      ii. Install the Shielded 50-pin to Serial/ELAN/TLAN Adapter for the Succession Media Card. See Procedure 10 on page 246.

4 Configure IP Line data on the Meridian 1 and Succession CSE 1000:

   a. Configure the IP address for the Meridian 1 and Succession CSE 1000 ELAN Ethernet interface. See Procedure 11 on page 248.

   b. Configure VoIP bandwidth management zones. See page 249.

   c. Configure IP Line physical TNs. See page 253.

   d. Configure virtual superloops. See page 259.

e. Configure Option 11C/11C-Mini mapping of virtual superloops. See page 260.

f. Configure Internet Telephone Meridian 1 and Succession CSE 1000 features. See page 263.

5 Configure IP Line data using OTM:

a. Manually add an IP Telephony node. See page 278.

b. Configure Voice Gateway Media Care properties. See page 282.

c. Configure DSP Profile data. See page 284.

d. Configure SNMP traps and Card routing table entries (if required). See page 288.

e. Configure Meridian 1 and Succession CSE 1000 Call Server ELAN IP (Active ELNK) address and TLAN Voice port on the Voice Gateway Media Card. See page 292.

f. Configure SNMP Community Names access for security. See page 295.

g. Configure access to the file server. See page 296.

h. Configure 802.1Q support and NAT support. See page 298.

6 Transmit Voice Gateway Media Card configuration data from OTM to the Voice Gateway Media Cards:

a. Set Leader 0 IP Address. See Procedure 24 on page 301.

b. Transmit node and card properties to Leader 0. See Procedure 25 on page 303.

c. Transmit card properties to all cards in the node. See Procedure 26 on page 305.

7 Upgrade card loadware and i2004 Internet Telephone firmware:

a. Verify card loadware and i2004 Internet Telephone firmware release. See page 309.

b. Determine latest loadware and firmware versions from Meridian 1 and Succession CSE 1000 Nortel Networks Web site. See page 309.

c. Upgrade Voice Gateway Media Card loadware (if required). See Procedure 28 on page 314.

d. Upgrade i2004 Internet telephone firmware (if required). See Procedure 30 on page 318.

8 Configure OTM alarm notification feature to receive IP Line SNMP traps. See Procedure 31 on page 323.

9 Assemble and install an Internet Telephone. Refer to the *Internet Terminals Description* (553-3001-217).

10 Change the default IP Line Command Line Interface (IPL>) Shell password. See Procedure 55 on page 415.

11 Configure the Internet Telephone Installer Passwords (see page 420).

a. Enable and set the administrative Internet Telephone Installer Password. See Procedure 58 on page 429.

b. If needed, enable and set a temporary Internet Telephone Installer Password. See Procedure 59 on page 433.

Succession CSE 1000 Rel 2.0 systems

Before you begin

1 Ensure that the system meets the following minimum requirements: Succession CSE 1000 Release 2.0

2 Upgrade the Succession CSE 1000 keycode to expand the ISM system limit to support the number of Internet Telephones you plan to install. Refer to *Software Input/Output: Maintenance* (553-3001-511) to use:

a. LD 22 command SLT to determine your Internet Telephone limit on the system.

b. LD 143 to expand the ISM parameters (if required).

3 Verify that you have access to the Signaling Server so Element Management can be run.

a. Check the Nortel Networks Software Downloads Web site to determine the latest software version and required patches (PEPs). See Procedure 111, "Downloading files from the Nortel Networks Web site" on page 719.

b. OTM 2.0 is be needed for certain IP Line 3.0 feature enhancements.

4 Ensure that browser caching is turned off so that Element Management runs properly. See "Internet Explorer browser configuration" on page 332.

5 Provision the IP network to support IP Telephony node and Internet Telephones.

a. Choose the Internet Telephone DHCP mode: Full, Partial, or None (static IP address). For Full DHCP mode, see "Configuring the DHCP server to support Full DHCP mode" on page 215.

b. Determine the Internet Telephone 10/100BaseT Ethernet LAN connection. There are two options for the connection:

   i. The connection can be a separate cable to the equipment closet.

   ii. The connection can be shared back to the closet using a desktop switch or the telephone's built-in 3-port switch.

c. Determine TLAN, ELAN IP address and Ethernet connections from the network IP administrator. See the "IP Network Engineering Guidelines" on page 133.

6 Check that the required LAN and WAN networking equipment and cables are installed. For IP networking requirements, see "IP Network Engineering Guidelines" on page 133.

Installation summary for Succession CSE 1000 Rel 2.0 systems

The following summary of steps can be used as a reference guide to install and configure an IP Telephony node and Voice Gateway Media Cards on Succession CSE 1000 Release 2.0 system. This summary is intended to serve as a pointer to the more detailed procedures contained in other chapters and to provide a sequential flow to the steps involved in the overall installation procedure.

Note: Complete all installation and configuration steps before you transmit data to the Voice Gateway Media Cards.

1 Complete the Voice Gateway Media Card installation summary sheet. See Table 45 on page 209.

2 Complete the Internet Telephone configuration data summary sheet. See Table 46 on page 210.

3 Install the hardware components:

a. Install the Voice Gateway Media Card(s). See Procedure 5 on page 230 for installing the ITG-P Line Cards and Procedure 7 on page 236 for installing the Succession Media Card.

b. Cable the Voice Gateway Media Cards:

i. Install the ELAN, TLAN, serial interface cable for the ITG-P Line Card. See Procedure 9 on page 243.

ii. Install the Shielded 50-pin to Serial/ELAN/TLAN Adapter for the Succession Media Card. See Procedure 10 on page 246.

4 Configure IP Line data on the Succession CSE 1000 Release 2.0:

a. Configure the IP address for the Succession CSE 1000 Release 2.0 ELAN Ethernet interface. See Procedure 11 on page 248.

b. Configure VoIP bandwidth management zones. See page 249.

c. Configure IP Line physical TNs. See page 253.

d. Configure virtual superloops. See page 259.

e. Configure Option 11C/11C-Mini mapping of virtual superloops. See page 260.

f. Configure Internet Telephone Meridian 1 and Succession CSE 1000 features. See page 263.

5 Configure IP Line data using Element Management:

a. Manually add an IP Telephony node. See page 338.

b. Configure SNMP traps and community names access for security. See page 343.

c. Configure DSP Profile data. See page 346.

d. Configure DiffServ Codepoint data, 802.1Q support, and NAT support. See page 351.

e. Configure Succession CSE 1000 Call Server ELAN IP (Active ELNK) address, TLAN Voice port, and the routing tables on the Voice Gateway Media Card. See page 353.

f. Configure file server access. See page 358.

g. Configure the loss plan. See page 360.

h. Configure Voice Gateway Media Care properties. See page 361.

6 Submit and transfer the node information to the Call Server. See page 364.

7 Transmit Voice Gateway Media Card configuration data the Voice Gateway Media Cards:

   a. Set Leader IP Address. See Procedure 43 on page 369.

   b. Transmit node and card properties to Leader. See Procedure 44 on page 372.

8 Upgrade card loadware and i2004 Internet Telephone firmware:

   a. Verify card loadware. See page 381.

   b. Verify card firmware release. See page 383.

   c. Download loadware and firmware files from the Nortel Networks Web site. See page 386.

   d. Upload the loadware and firmware files to the file server. See page 387.

   e. Upgrade the loadware on the Voice Gateway Media Card. See page 389.

   f. Reboot the card. See page 393.

   g. Upgrade the firmware on the card. See page 394.

9 Configure OTM alarm notification feature to receive IP Line SNMP traps. See Procedure 31 on page 323.

10 Assemble and install an Internet Telephone. Refer to the *Internet Terminals Description* (553-3001-217).

11 Change the default IP Line Command Line Interface (IPL>) Shell password. See Procedure 55 on page 415.

12 Configure the Internet Telephone Installer Passwords (see page 420).

   a. Enable and set the administrative Internet Telephone Installer Password. See Procedure 58 on page 429.

   b. If needed, enable and set a temporary Internet Telephone Installer Password. See Procedure 59 on page 433.

Voice Gateway Media Card installation summary sheet

Nortel Networks recommends that you complete a Voice Gateway Media Card installation summary sheet (Table 45 on page 209) as you unpack, inventory, and provision the cards. IP address information is normally supplied by the your IP Network Administrator.

To complete the installation summary sheet, you need to know:

- The MAC address. This is the motherboard Ethernet address on the Voice Gateway Media Card faceplate sticker (for example 00:60:38:01:12:77).
- The ELAN Management IP address of the motherboard Ethernet interface used to perform management through OTM and to communicate with the Meridian 1 and Succession CSE 1000.
- The TLAN Node IP address for the IP Telephony node.
- The TLAN card IP address of the voice interface on each card.
- The IP address of the active ELNK Ethernet interface on the Meridian 1 and Succession CSE 1000 core.

Nortel Networks recommends that you complete an Internet Telephone configuration data summary sheet (Table 46 on page 210) as you install and configure Internet Telephones.

Table 45
Voice Gateway Media Card installation summary sheet

Site_____ **Meridian 1 or Succession CSE 1000 system**_____
Meridian 1 and Succession CSE 1000 customer_____
Node ID (Number)_____
TLAN Node IP address_____
Meridian 1 and Succession CSE 1000 active ELNK IP address_____
SNMP Manager List IP addresses_____
TLAN gateway (router) IP address_____
TLAN subnet mask_____
ELAN gateway (router) IP address_____
ELAN subnet mask_____

TN	ELAN Management MAC address	ELAN Management IP address	TLAN (Voice) Card IP address	Card role
				Leader
				Follower (OTM: Leader1)
				Follower
				Follower
				Follower
				Follower
				Follower
				Follower
				Follower
				Follower
				Follower
				Follower

Internet Telephone configuration data summary sheet

Table 46
Internet Telephone configuration data summary sheet

No DHCP								
			Partial DHCP					
						Full DHCP		
IP address	subnet mask	Gateway IP address	Connect server IP address*	Node#	VTN	DN	User Name	User Location
*Connect server IP address is the Node IP address of the IP Telephony node.								

Configuration of the DHCP Server

Contents

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Format for Nortel Networks Internet Telephone DHCP Encapsulated Vendor Specific Option 218
Format for Nortel Networks Internet Telephone DHCP Site Specific Option. 222

Reference list

The following are references for this section:

- *Internet Terminals Description* (553-3001-217)

Overview

This chapter provides general guidelines to configure a host with a Dynamic Host Configuration Protocol (DHCP) server to support the i2002 Internet Telephones, i2004 Internet Telephone, and i2050 Software Phone.

Note 1: This chapter assumes that you are familiar with RFC 2131 "Dynamic Host Configuration Protocol", RFC 1533 "DHCP Options and BOOTP Vendor Extensions", and the Help manual for the DHCP server on your host. A convenient source for RFCs is http://www.ietf.org/.

Note 2: For a general overview of DHCP server technology, refer to Appendix E on page 677.

Note 3: For DHCP server setup and configuration information, refer to Appendix F on page 697.

i2002 and i2004 Internet Telephone, and i2050 Software Phone

The i2002 Internet Telephone, i2004 Internet Telephone, and the i2050 Software Phone are Voice-over Internet Protocol (VoIP) telephones that function as a terminal to the Meridian 1 and Succession Communication Server for Enterprise 1000. The Internet Telephone encodes voice as binary data and packetizes the data for transmission over an IP Network to the Voice Gateway Media Card or to another Internet Telephone.

The Nortel Networks Internet Telephone can act as a DHCP client in one of two modes:

- Partial DHCP mode
- Full DHCP mode

Partial DHCP mode

When the Internet Telephone is configured to operate in partial DHCP mode, the DHCP server needs no special configuration to support Internet Telephones. The Internet Telephone receives the following network configuration parameters from the DHCP server:

- IP address configuration for the Internet Telephone
- Subnet mask for the Internet Telephone IP address
- Default gateway for the Internet Telephone LAN segment

Full DHCP mode

In full DHCP mode, the DHCP server requires special configuration. The Internet Telephone obtains network configuration parameters and connect server configuration parameters from specially configured DHCP servers.

The following configuration parameters are provided for the primary and secondary connect servers:

- Connect server IP address. For IP Line, the connect server IP address is the IP Telephony node IP address.
- A port number of 4100.
- A command value of 1 that identifies the request to the connect server as originating from an Internet Telephone.
- A retry count typically equal to 10.

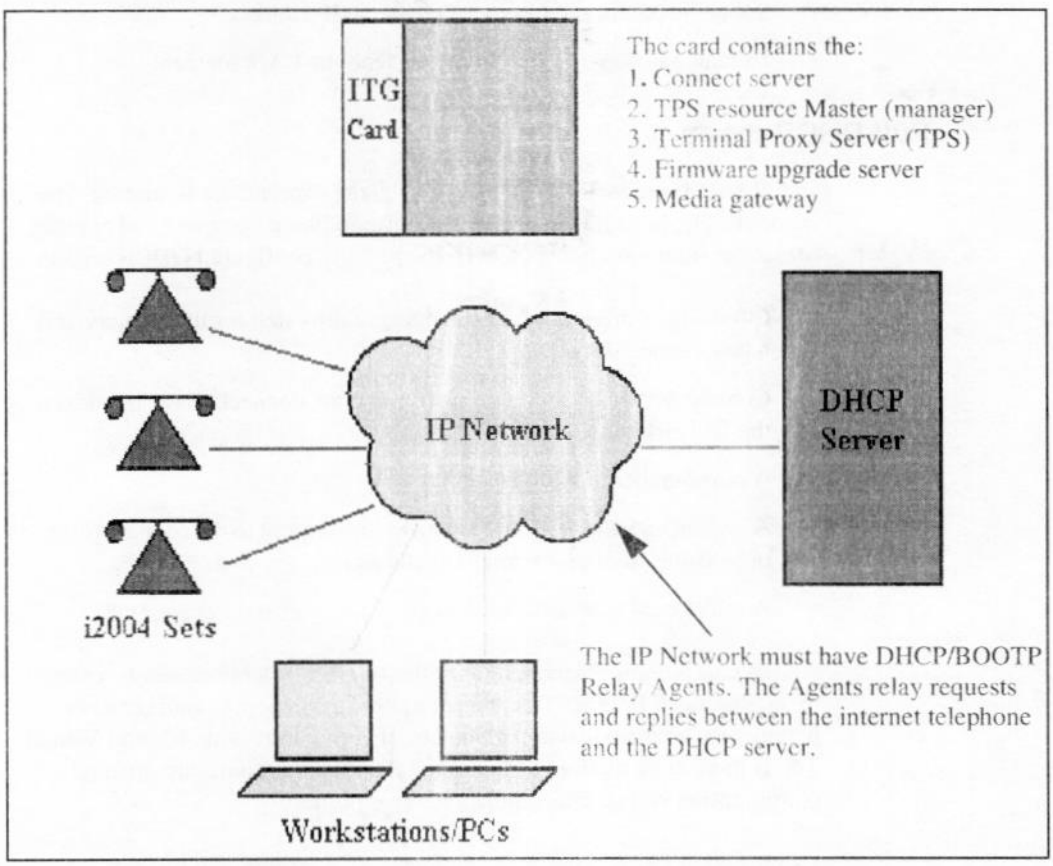
All the configuration parameters for the Internet Telephone can be entered manually. Each Internet Telephone requires the network configuration parameters, connect server parameters, IP Telephony node ID, and Virtual TN. If there are a number of Internet Telephones to configure, manual configuration is time consuming and error prone.

Using Full or Partial DHCP to automatically configure the Internet Telephones is more efficient and flexible. This ensures that current information is used.

Note 1: The IP Telephony node ID and Virtual TN must always be configured manually even in Full DHCP mode.

Note 2: In Partial DHCP mode the connect server parameters, node ID and Virtual TN must be entered manually.

Figure 11
DHCP block diagram



802.1Q Configuration of the i2002/i2004 Internet Telephones

The 802.1Q VLAN support is configured from the user display interface of the i2002/i2004 Internet Telephones. This configuration takes place during the initial configuration procedure of the telephone.

For the 802.1Q configuration procedures of the Internet Telephones, see *Internet Terminals Description* (553-3001-217).

Configuring the DHCP server to support Full DHCP mode

The DHCP capability feature of the Internet Telephone enables the telephone to receive network configuration parameters and specific connect server parameters. This section describes the Internet Telephone's unique class identifier and requested network configuration and connect server parameters for automatic configuration.

Internet Telephone class identifier

The Internet Telephone is designed with a unique class identifier that the DHCP server can use to identify it. All Nortel Networks Internet Telephones use the same text string, "Nortel-i2004-A". The ASCII string is sent inside the Class Identifier option of the Internet Telephone's DHCP messages.

The DHCP server also includes this string in its responses to the Internet Telephone DHCP client. This makes it possible to notify the Internet Telephone that the server is Internet Telephone-aware, and that it is safe to accept the server's offer. This string appears in the beginning of a list of specific Voice Gateway Media Card information that the Internet Telephone DHCP client requests.

When the DHCP server is configured to recognize the Internet Telephone as a special class, the DHCP server can treat the Internet Telephone differently than other DHCP clients. DHCP host configuration parameters can then be grouped by class and only information relevant to the Internet Telephone DHCP client, such as the connect server parameters, is supplied.

Also, the administrator can design the network according to the client's class, if necessary, making maintenance easier. Depending on the capabilities and limitations of the DHCP server used and the design of the network, some of these advanced functions are not available.

Requested Network Configuration Parameters

Nortel Networks Internet Telephones, using Full DHCP mode, can be configured automatically by an Internet Telephone-aware DHCP server by requesting a list of connect server configuration parameters. The Internet Telephone uses DHCP, an industry standard protocol, to request and receive the information.

The Internet Telephones operating in Partial DHCP mode can receive an IP address from any DHCP server. In Full DHCP mode, the server must be configured to respond to the request for the vendor specific encapsulated options.

Table 47 lists the network configuration parameters requested by the Internet Telephone in the Parameter Request List option (Option Code 55) in the DHCPDISCOVER and DHCPREQUEST messages. The DHCPOFFER and the DHCPACK reply messages from the DHCP server must contain the options in Table 47.

Table 47
Internet telephone network configuration requirements (Part 1 of 2)

Parameter Request (Option Code 55)	DHCP Option Code
Subnet mask - the client IP subnet mask.	1
Router/Gateway(s) - the IP address of the client's default Gateway. (Not required in DHCPOFFER in Internet Telephone Firmware 1.25 and later for compatibility with Novell DHCP server.)	3
Lease Time - implementation varies according to DHCP server.	51

Table 47
Internet telephone network configuration requirements (Part 2 of 2)

Parameter Request (Option Code 55)	DHCP Option Code
Renewal time - implementation varies according to DHCP server.	58
Rebinding interval - implementation varies according to DHCP server.	59
IP Line Site Specific or Vendor Specific encapsulated/site options.	43, 128, 144, 157, 191, 251

The first five parameters in Table 47 are standard DHCP options and have predefined option codes. The last parameter is for Voice Gateway Media Card information, which does not have a standard DHCP option. The server administrator must define a vendor encapsulated and/or site specific option to transport this information to the Internet Telephone.

This non-standard information includes the unique string identifying the Internet Telephone and the connect server parameters for the primary and secondary servers. The Internet Telephone must receive the connect server parameters to connect to the IP Telephony node.

The administrator must use one of the five site specific or vendor encapsulated option codes to implement the Voice Gateway Media Card information. This user-defined option can then be sent as is, or encapsulated in a Vendor Encapsulated option with Option Code 43. The method used depends on the DHCP server's capabilities and what options are already in use by other vendors.

The Internet Telephone rejects any DHCP Offers/Acks that do not contain:

- A Router option. The Internet Telephone requires a default gateway (router).
- A Subnet Mask option.
- Either a Vendor Specific option (see Note 1) or a Site Specific option (see Note 2).

Note 1: The Vendor Specific Option is 43. Windows NT DHCP Server (up to SR4) supports only 16 octets of data for the vendor-specific option, which is insufficient to support the minimum length of the Internet Telephone-specific string. If you use a Windows NT DHCP Server, you must select the Site Specific option to accommodate the Internet Telephone-specific string.

Note 2: The Site Specific options are all DHCP options between 128 (0x80) and 254 (0xFE). These options are reserved for Site Specific use by the DHCP RFCs.

Format for Nortel Networks Internet Telephone DHCP Class Identifier Option

All Nortel Networks Internet Telephones fill in the Class ID option of the DHCP Discovery and Request messages with the null-terminated, ASCII-encoded string Nortel-i2004-A, where A identifies the version number of the information format of the Internet Telephone.

The Class Identifier Nortel-i2004-A must be unique in the DHCP server domain.

Format for Nortel Networks Internet Telephone DHCP Encapsulated Vendor Specific Option

The following definition describes the Nortel specific, Encapsulated Vendor Specific option for the i2002 and i2004 Internet Telephones, and i2050 Software Phone. This option must be encapsulated in a DHCP Vendor Specific Option (refer to RFC 1533) and returned by the DHCP server as part of each DHCPOFFER and DHCPACK message for the Internet Telephone to accept these messages as valid. The Internet Telephone extracts the relevant information from this option and uses it to configure the connect server IP address, the port number (4100), a command value of one, and retry count for the primary and secondary connect servers.

Either this encapsulated vendor specific option or a similarly encoded site specific option must be sent. The DHCP server must be configured to send one or the other but not both. The choice of using either the Vendor Specific or the Site Specific option is provided to enable Windows NT DHCP servers to be used with the Internet Telephone (Windows NT servers do not properly implement the Vendor Specific Option, and as a result, Windows NT implementations must use the Site Specific version).

Format of the option

The format of the Encapsulated Vendor Specific option is Type, Length, and Data as shown below.

Type (1 octet):

There are five choices:

- 0x80 (Site Specific option 128)
- 0x90 (Site Specific option 144)
- 0x9d (Site Specific option 157)
- 0xbf (Site Specific option 191)
- 0xfb (Site Specific option 251)

Providing a choice of five types enables the Internet Telephone to work in environments where the initial choice could already be in use by a different vendor. Pick only one value for TYPE byte.

Length (1 octet)

The Length value is variable. Count only the number of octets in the data field (see below).

Data field (variable number of octets)

The data field contains an ASCII-encoded character string that can be optionally null-terminated. This string can be NULL terminated, although the NULL is not required for parsing.The string is:
"Nortel-i20xx-A,iii.jjj.kkk.lll:ppppp,aaa,rrr;iii.jjj.kkk.lll:pppp,aaa,rrr."
where the parameters for the data field are outlined in Table 48 on page 220.

Table 48
Data field parameters

Parameter	Description
Nortel-i2004-A	Uniquely identifies that this is the Nortel option and is a response from a server that can provide the correct configuration information to the i2002 and i2004 Internet Telephones, and the i2005 Software Phone.
ASCII Characters Comma (,) Colon (:) Semicolon (;) Period (.)	ASCII "," separates fields ASCII ":" separates the IP address of the bootstrap server node IP from the Transport Layer port number. ASCII ";" separates the Primary from Secondary bootstrap server information. The bootstrap server is the Active Leader of the IP Telephony node. ASCII "." signals end of structure
iii.jjj.kkk.lll:ppppp	Identifies IP address and port number for server (ASCII encoded decimal)
aaa	Identifies Action for server (ASCII encoded decimal, range 0..255)
rrr	Identifies retry count for server (ASCII encoded decimal, range 0..255)

1 "aaa" and "rrr" are ASCII encoded decimal numbers with a range of 0..255. They identify the "Action Code" and "Retry Count", respectively, for the associated TPS server. They are stored as 1 octet (0x00..0xFF) in the Internet Telephone. These fields must be no more than three digits long.

2 The first server is always considered "Primary"; the second server always considered "Secondary".

3 If only one server is required, terminate primary TPS sequence immediately with "." instead of ":" for example, "Nortel-i20xx-A,iii.jjj.kkk.lll:ppppp,aaa,rrr."

4 Valid options are one or two servers (0 or 3 is not allowed). However, it is recommended that two servers be used. For i2004 Internet Telephone firmware version 3002B00, the valid option is two servers.

Note: If there is only one connect server (that is, only one IP Telephony node is configured), enter the same information for server 1 and server 2.

5 Action code values:

- **a** 0 – reserved
- **b** 1 – UNIStim Hello (currently this type is the only valid choice)
- **c** 2..254 – reserved
- **d** 255 – reserved

6 iii,jjj,kkk,lll are ASCII encoded, decimal numbers representing the IP address of the server. They do not need to be three digits long as the **.** and **:** delimiters guarantee parsing. For example, '001', '01', and '1' would be parsed correctly and interpreted as value 0x01 internal to the Internet Telephone. These fields must be no more than three digits long.

7 ppppp is the port number in ASCII encoded decimal. It does not need to be five digits long as the **:** and **,** delimiters guarantee parsing. For example, '05001', '5001', '1', '00001' would be parsed correctly and accepted as correct. The valid range is 0-65535 (stored internally in the Internet Telephone as hexadecimal in range 0..0xFFFF). This field must be no more than five digits long.

8 In all cases, the ASCII encoded numbers are treated as decimal values and all leading zeros are ignored. Specifically, a leading zero does not change the interpretation of the value to be OCTAL encoded. For example, 0021, 021, and 21 are all parsed and interpreted as decimal 21.

Format for Nortel Networks Internet Telephone DHCP Site Specific Option

The following definition describes the Nortel specific, Site Specific option for the i2002 and i2004 Internet Telephones, and i2050 Software Phone. This option uses the "reserved for site specific use" DHCP options (128 to 254 - refer to RFC 1541 and RFC 1533) and must be returned by the DHCP server as part of each DHCP OFFER and ACK message for the Internet Telephone to accept these messages as valid.

The Internet Telephone retrieves the relevant information and uses it to configure the IP address for the primary and (optionally) secondary TPSs. Either this site specific option must be present or a similarly encoded vendor specific option must be sent (as previously described), that is, configure the DHCP server to send one or the other but not both. The choice of using either Vendor Specific or Site Specific options enables Windows NT DHCP servers to be used with the Internet Telephone. Windows NT servers do not properly implement the Vendor Specific Option and as a result, Windows NT implementations must use the Site Specific version.

Format of the option

The format of the field is Type, Length, Data. The format of the Site Specific Option is the same as the Encapsulated Vendor Specific Option (see "Format of the option" on page 219).